VoIP Phone

Model EP-8201

User Manual

Release 1.2

1

Overview

The EP-8201 VoIP Phone is fully compatible with the open SIP industry standards. This feature-rich VoIP Phone is designed as an enterprise grade VoIP device to work seamlessly with most of the existing SIP systems.

Figure 1-1 *EP-8201 VoIP Phone*

1.1 VoIP Phone Features

The EP-8201 VoIP Phone has the following features:

- Standard SIP
- Standard Protocols: TCP/UDP/IP, RTP, HTTP, HTTPS, ARP, DNS, DHCP, NTP/SNTP, FTP, TFTP, and SSL
- VoIP Speech codecs: including G.711 a-law and u-law, GSM, G.723.1, G.729a/b/ab
- Interoperable with various 3rd party VoIP end user devices, Proxy/Registrar/Server, and gateway products
- Up to 4 SIP registrations (same server or different servers) with BASIC or DIGEST authentication (MD5, MD5-sess)
- Full-duplex Speakerphone with echo cancellation
- Handsfree operation via speakerphone or headset mode
- Standard phone features: Caller ID Display, Call Waiting, Call Hold, Call Transfer, Call Forward, Call Conference, Call History, Speed Dial, Phonebook
- Special phone features: Dial Plan, Hotline, Voice Mail
- DTMF modes: In-band and out-of-band DTMF (RFC2833/SIP INFO) dialing
- Dial Plan
- 128 x 64 dot matrix LCD (sixe: 4 cm x 7 cm) with backlight and 4-level contrast controls
- 4 ring types, 6 ringing levels (including silence level)
- 6-level volume controls for both speakerphone and headset modes
- DHCP Client, Static IP, PPPoE support for the LAN Port
- Built-in network switch or router for the PC Port
- Backup Server support for Single Server Mode
- Jitter Buffer, VAD, CNG and PLC
- Redundant DNS support
- QoS Support (ToS / VLAN)
- Built-in Web Server for Device Configuration
- Built-in Phone Menu
- Auto Firmware Upgrade and Phonebook Update
- Auto Provisioning via http/ftp/tftp

1.2 Front View

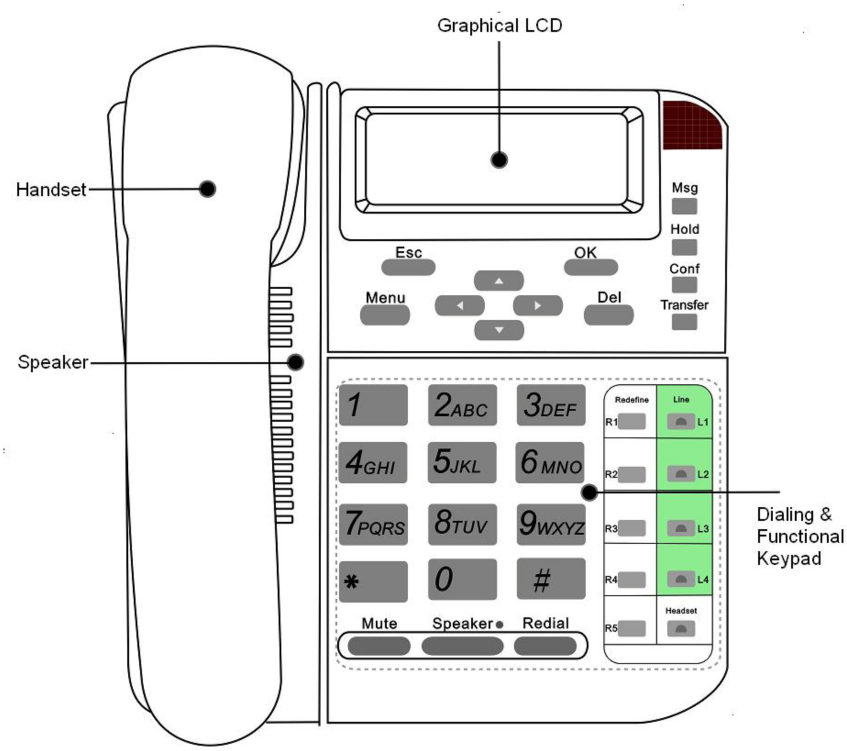


Figure 1-2 EP-8201 Front View

1.3 Bottom View

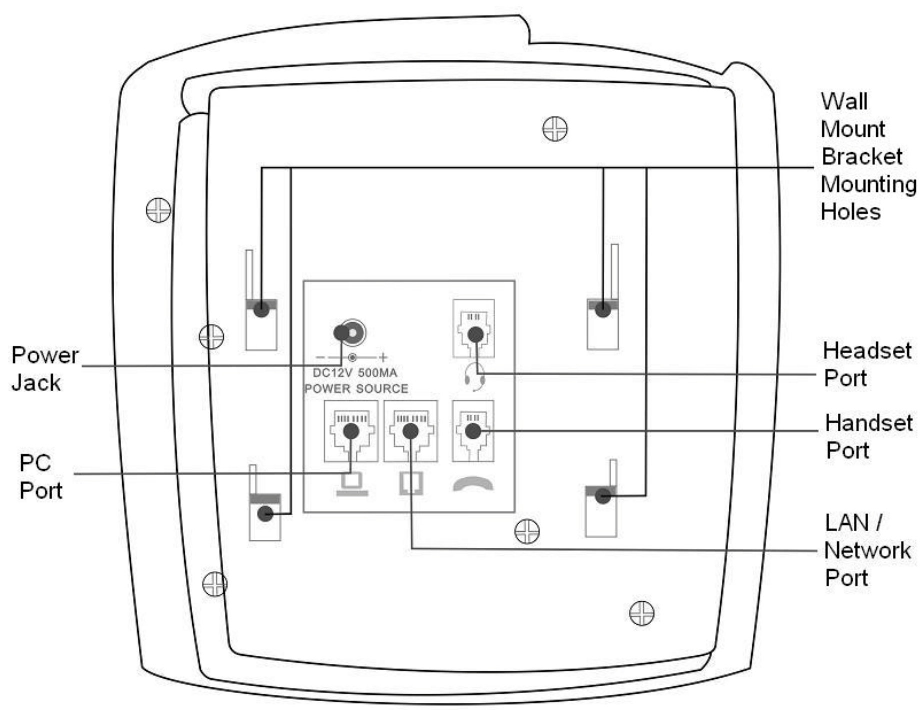


Figure 1-3 *EP-8201 Bottom View*

1.4 Structure

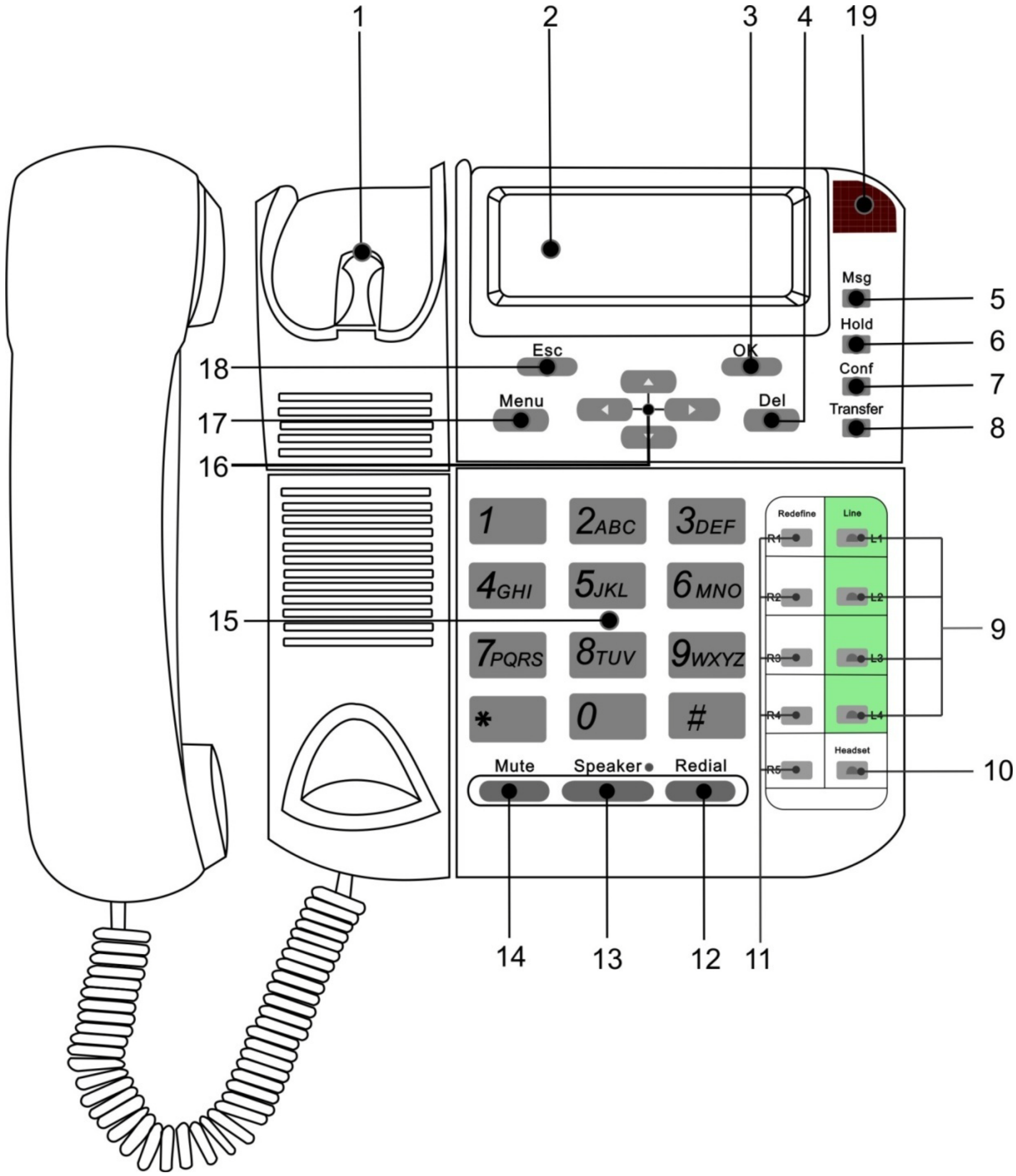


Figure 1-4 *EP-8201 Structure*

1 Hook Switch

2 LCD

3 OK key

4 Del key (Delete)

5 Msg key (Message)

6 Hold key

7 Conf. key (Conference)

8 Transfer key

9 Line keys + LED (L1/L2/L3/L4)

10 Headset key + LED

11 Redefine keys (R1/R2/R3/R4/R5)

12 Redial

13 Speaker key + LED

14 Mute key

15 Dialing Keypad

16 Navigator keys (UP/ DOWN/LEFT/RIGHT)

17 Menu key

18 ESC key (Escape)

19 Visual Indicator (Red LED)

2

Installation

Please follow the steps below to prepare and install the EP-8201.

1. Unpacking the EP-8201 gift box.
2. Checking the contents as described in Section 2.1.
3. Connecting the EP-8201 as described in Section 2.2.
4. Powering up the EP-8201.
5. Acquiring EP-8201 Phone IP address.
6. Configuring EP-8201 via Web Browser.
7. Configuring EP-8201 via Phone Menu (Basic settings only).

2.1 Package Contents

EP-8201 is shipped with the following items as shown in Figure 2-1 below:

1. One EP-8201 main Boad
2. One Handset
3. One Coild Handset cord
4. One Base Stand
5. One Ethernet Cable
6. One AC/DC Adapter - 12VDC/500mA Output (Optional)

   Note: This is not required if your network switch supports POE.

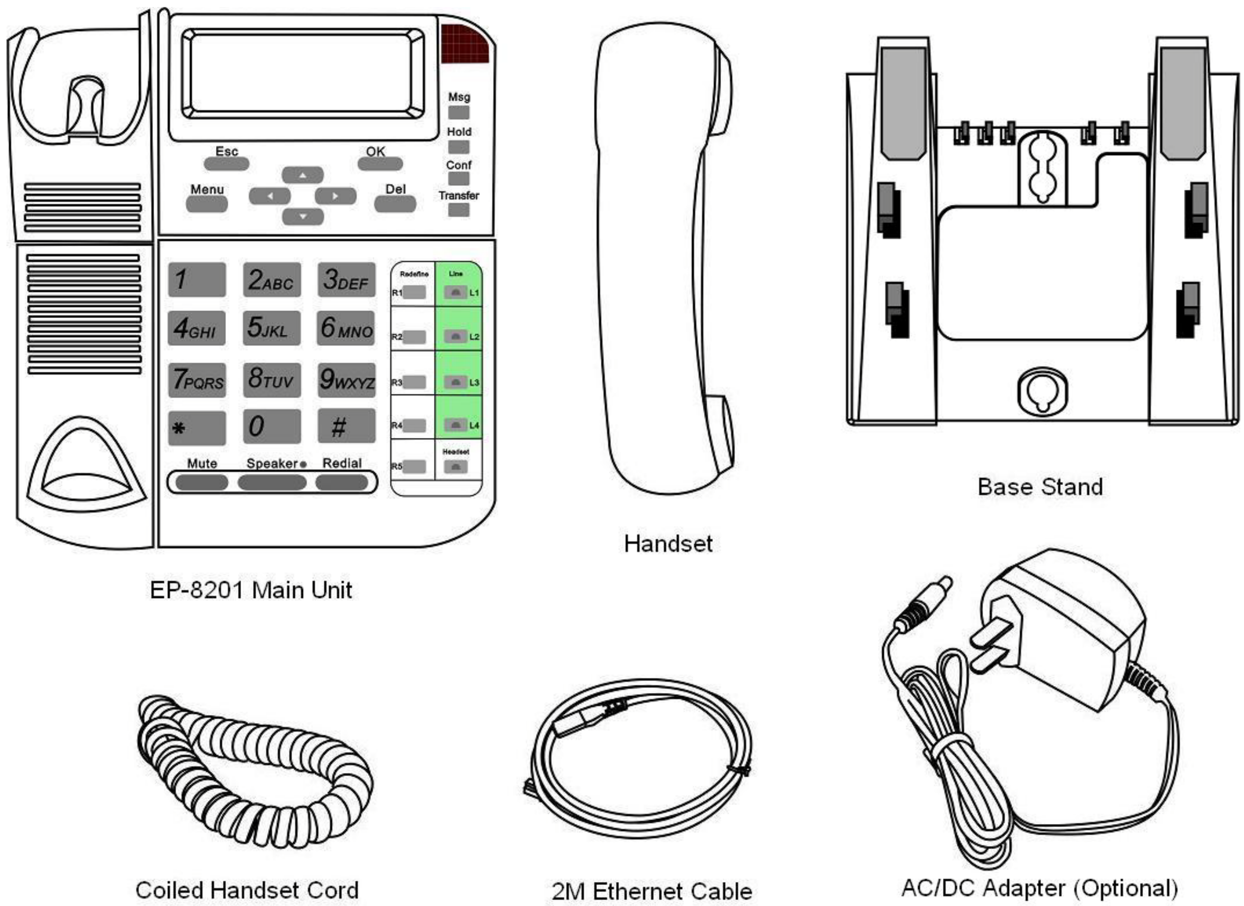
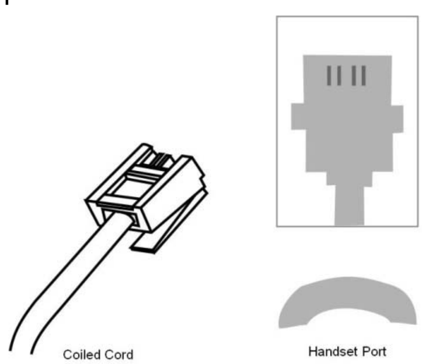


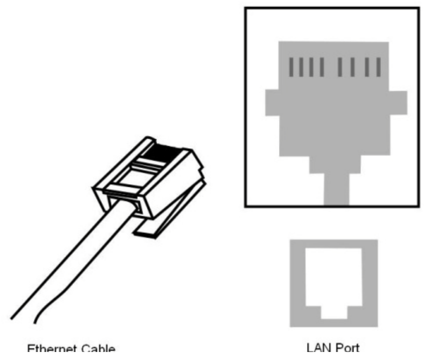
Figure 2-1 EP-8201 Package Content

2.2 Setting Up EP-8201

1. Connect the coiled telephone cord to handset and the base of the VoIP Phone.



2. Use the enclosed (or equivalent) Ethernet cable to connect the LAN port to a hub or switch, or to a DSL Router or Cable Modem.

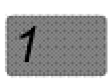


3. If network does not support PoE, connect the power supply to the DC Jack on the bottom of the EP-8201 (Use the one enclosed or equivalent).

4. For the first time installation, the phone will scan the network for available services which are DHCP, PPPoE, and Fixed IP. Select the preferred service and enter the required information if needed. No user input is required for DHCP service. User ID and Password are required for PPPoE. IP address, Netmask, and Gateway IP Address are required for Fixed IP mode.

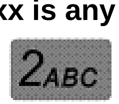
2.3 Acquiring Network Port IP Addresses

Once the EP-8201 is properly setup and powered up, the IP Addresses assigned to the LAN port and PC port can be retrieved via the Phone Menu as described below.

- Press  .
- Press  (System Tools).
- Press  (Phone Status).
- Press  (LAN Port) to view the LAN port IP.

  The phone LCD displays:
  LAN PORT
  - **STATUS: WORKING.**
  - **IP: xxx.xxx.xxx.xxx**

  where xxx is any valid IP address between 0 and 255.

- Or Press  (PC Port) to view the PC port IP. If the PC port is set to Bridge Mode, there is no IP address assigned. In this case, a PC connected to this port is in the same network segment as the LAN port.

2.4 Accessing the Built-in Web Server

The built-in Web Server can be accessed by entering the LAN / PC IP address in a web browser. Please see below to determine which IP Address to be used to access the built-in Web Server.

Use LAN Port IP address when:

1. A PC and the LAN port are connected to the same network segment. This condition applies to a PC connected to the PC port when it is set to Bridge Mode.
2. LAN IP address is public and the PC has internet access.
Note: If a private IP is assigned to the LAN port, it may still be accessed from the internet provided that the local router is set up properly. Please consult your network administrator for more information.

Use PC Port IP address when:

1. The PC Port is set to Fixed IP mode and a PC is connected to the same network segment. Please make sure that the PC IP is setup properly.

To access the built-in Web Server, type the proper IP address (for example: *192.168.2.124* or *http://192.168.2.124* in the address field of a Web Browser (IE, Firefox, etc.).

Figure 2-2 Entering Phone IP Address to a Web Browser

Once the EP-8201 responds to the HTTP request, the Web Browser will prompt for a login window as shown below.

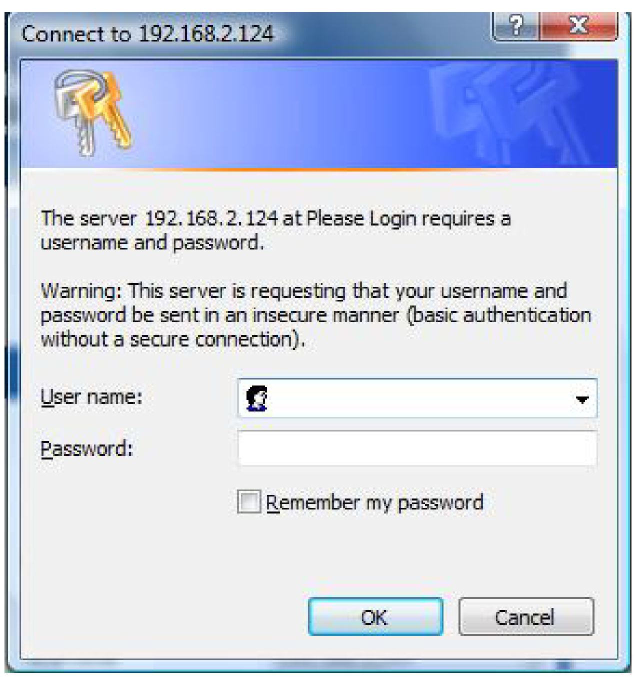


Figure 2-3 Web Browser Authentication Window

EP-8201 supports two login levels.

For Administrator, please enter User name = “**admin**” and Password = “**admin**” (factory default).

For User, please enter User name = “**user**” and the Password = “**1234**” (factory default).

Both passwords can be changed in the Administrator mode. Only user password can be changed in the User mode. Please keep a record of the new passwords if changed. There is a Star Command to reset the passwords to the factory defaults (Please see section x.xx for more information).

The Administrator mode allows full access to the built-in Web Server whereas the User mode restricts the user from accessing the **Call Settings** page.

3

Web Configuration

Once the login is successful, the Web Browser brings up the **Status** page as shown below. The built-in Web Server is divided into five categories: **Status**, **Configurations**, **Phone Book**, **Tools**, and **Logout**. They can be access by clicking on the left hand menu column.

Figure 3-1 EP-8201 Built-in Web Server – Status Page

4.1 Status Page

The Status page provides a brief summary on the Phone (Device) and Network information.

Phone Information

a) **Profile X Number**
Up to 4 Profiles can be defined; each profile contains one phone registration to the same or different server.

b) **Profile X Status**
This field shows the status of server registration for each profile. If the device registers to the designated server successfully, it displays “LOGIN”; otherwise, it displays “LOGOUT”. If the profile is not defined, it display “NO CONFIG”.

c) **Serial Number**
Each EP-8201 is assigned with a unique serial number by the factory. This number is important for auto provision, technical support, and warranty service. This serial number is also printed on the product label at the bottom.

d) **Firmware Version**
This field identifies the current Firmware Version installed.

e) **Hardware Model**
This field identifies the hardware model and version.

Network Information

a) **LAN Port**
This field shows IP address assigned to the LAN port.

b) **PC Port**
This field shows IP address assigned to the PC port.

c) **PPPoE**
This field shows the dial up status when PPPoE is enabled for ADSL login.

d) **Default Route**
The Default Route shows the IP address of the default gateway / router that is used in the current network environment.

e) **DNS Server**
This field shows the IP address of the DNS server to be used for domain name interpretation.

4.2 Configuration Page

To access the **Configurations** page, click on the “Configurations” tab on the left hand column. This brings up all the pages under this tab: **Preference**, **Network**, **Call Settings**, and **Phone Settings**.

3.2.1 Preference

This page configures the general settings in the device: **Language**, **Time Zone**, **Display SIP Server Time**, **Time server**, **Time Format**, **Auto-Provision**, **Key(#)**, **Phone Book Mode**, **Network Tones**, **Speaker Phone Mic Input Gain**, and **Central Management**.

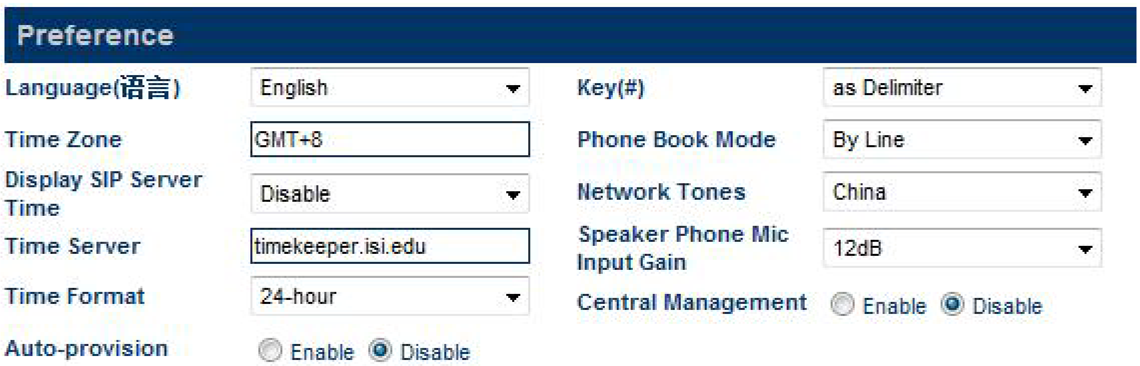


Figure 1-1 Configurations Page – Preference

a) **Language** - This field sets the language to be used for initial access to the built-in Web Server. The languages currently available for selection are **English** and 简体中文(Simplified Chinese). Once the language change is saved, it does not take effect until the web server restarts.

Figure 1-2 Webpage Language Selection

To change the display language immediately, you can select the language icon located at the top right hand corner (as shown below). However, this does not change the default language.

Figure 1-3 Viewing Language Selection

b) **Time Zone** – This parameter specifies your local time zone in order for the date/time to be correctly displayed since the date/time obtained from a network time server is referenced to the Greenwich Mean Time (GMT). If your time zone is 8 hours ahead of the GMT, you need to enter the value "GMT+8" in this field.

c) **Display SIP Server Time** – This parameter enables the phone to receive the data/time information from the designated SIP Server. If this is enabled, the **Time Server** parameter below will be disabled automatically.

d) **Time Server** – This parameter specifies the location of the network time server for obtaining the date and time information. It accepts both domain name and IP address. If the **Display SIP Server Time** is enabled, this parameter is disabled automatically.

e) **Time Format** – This specifies the display time format (12-hour or 24-hour).

f) **Auto Provision** – This parameter enables or disables the Auto Provision procedures. The **Auto Provision** is a batch script to obtain configuration and firmware upgrade information from a server. Once this option is enabled, two additional parameters (**Provision Server** and **Provision Interval**) are displayed. The **Provision Server** specifies the location of the designated provision server. The auto provision procedure is executed at boot up time and is repeated at a duration specified in the parameter **Provision Interval**.

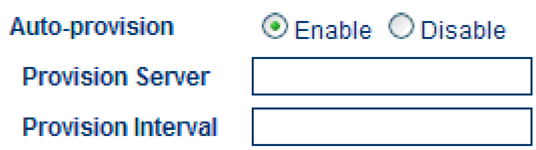


Figure 1-4 Auto Provision Setting

g) **Key(#)** – When dialing a VoIP number, the VoIP device needs to wait for the user to c8omplete the number dialing before the call request is actually sent to the server. This parameter enables or disables the "#" key to be used to signal the number dialing is completed and the call request can be execute immediately.

h) **Phone Book Mode** – This parameter defines if one phone book is used for all four profiles (**Global**) or for each profile (**By Line**).

i) **Network Tones** – This parameter defines the network tones to be used. The predefined networks tones are: **China**, **Hong Kong**, **Taiwan**, **New Zealand**, **United Kingdom**, **United States**, **Korea, Slovenia**, **Czechoslovakia**, **India**, **Singapore**, **Israel**, **Malaysia**, **Indonesia**, **Thailand**, **Romania**, **Bangladesh**, and **Customized**. The **Customized** option allows user to define his own network tones. If the desired network tones selection is not available, user can use this **Customized** option.

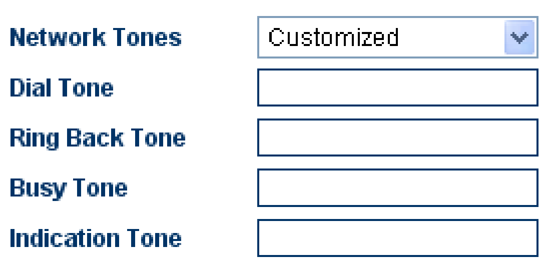


Figure 1-5 Network Tones Setting

Each network tone is defined as

nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4,
where **nc** is the number of cadences
rpt is the repeat counter(0 - infinite, 1~n - repeat 1~n times)
c1on is the cadence one on duration (in milliseconds)
c1off is cadence one off duration (in milliseconds)

c2on is the cadence two on duration (in milliseconds)
c2off is the cadence two off duration (in milliseconds)
c3on is the cadence three on duration (in milliseconds)
c3off is the cadence three off duration (in milliseconds)
f1 is the tone #1, 300-3000(Hz)
f2 is the tone #2, 300-3000(Hz)
f3 is the tone #3, 300-3000(Hz)
f4 is the tone 34, 300-3000(Hz)
p1 is the attenuation index for tone #1, 0~31(0=3dB, -1dB increments)
p2 is the attenuation index for tone #2, 0~31(0=3dB, -1dB increments)
p3 is the attenuation index for tone #3, 0~31(0=3dB, -1dB increments)
p4 is the attenuation index for tone #4, 0~31(0=3dB, -1dB increments)

Two examples of network tone definition are shown below.

1. A New Zealand Dial Tone (400 Hz) is defined as **0,0,0,0,0,0,0,0,400,0,0,0,10,0,0,0**.
2. A New Zealand Busy tone (400Hz with a cadence of 500ms on and 500ms off (repeat)) is defined as **1,0,500,500,0,0,0,0,400,0,0,0,10,0,0,0**.

j) **Speakerphone Mic Input Gain** – This parameter is intended to tune the Speakerphone performance. Increase the Mic gain if the volume heard at the other party is low. Reduce the Mic gain if the echoes or howling occurs locally.

k) **Centralized Management** – This is a proprietary protocol to support remote management of the EP-8201. Please consult your local agent or representative for more information.

3.2.2 Network Configuration

This page configures the network interface for the **LAN Port** and the **PC Port**.

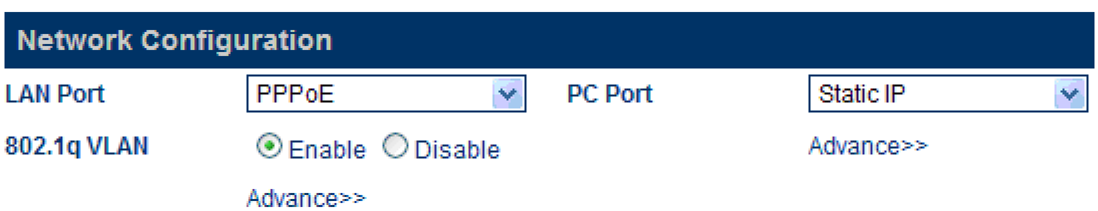


Figure 1-6 Network Configuration

LAN Port

The LAN port is intended for internet access. It is normally connected to a network device (router or ADSL modem) for internet access. Three operational modes are supported.

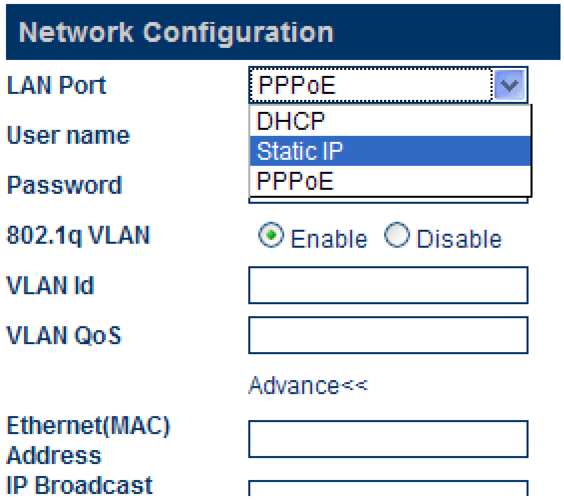


Figure 1-7 LAN Port Setting (Mode Selection)

- **DHCP** – This mode should be selected If the network device functions as a DHCP host, This allows the HT-812P to obtain all related network information / settings from the DHCP host.

- **Static IP** – This mode sets the LAN port IP manually which can either be a public or private IP. Other network settings (Subnet Mask, Default Route, Primary DNS, Secondary DNS) should also be entered accordingly.

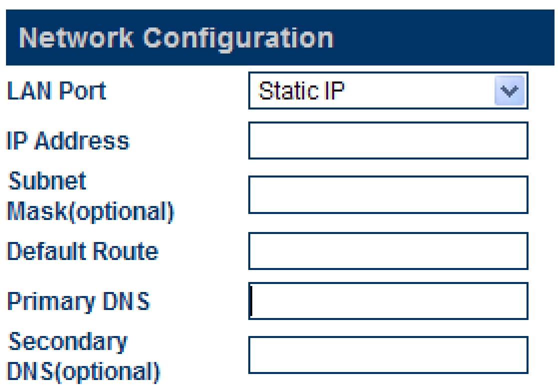


Figure 1-8 LAN Port Setting (Static IP)

- **PPPoE** – This selection is intended for broadband connection (ADSL / Cable modem) that requires dial up / authentication using PPPoE protocol. Both **User Name** and **Password** are required. Please consult your service provider for more information if needed. One advantage with the PPPoE dial up is that the IP address obtained for the LAN port is normally a public IP.

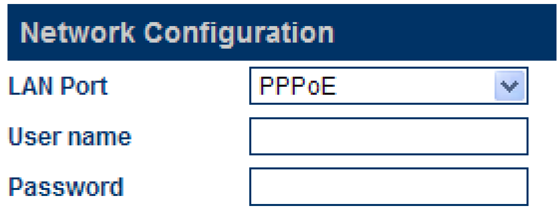


Figure 1-9 LAN Port Setting (PPPoE)

More advanced parameters for **802.1q VLAN** and **MAC** settings are available. Please consult your network administrator for assistance if needed.

PC PORT

The PC port is intended to provide an Ethernet connection to other network devices (for example: PC, network HUB.). Two modes of operation are available:

1. **Bridge mode** - This mode allows the network traffics at the PC port to be bypassed to LAN port. This means that the network device share the same network segment as the LAN port. There is no IP address assigned to the PC port.
2. **Fixed IP** - This mode sets the PC port **IP Address** (private IP) and **Subnet Mask** manually. This creates a new network segment for the network devices connected to the PC Port.

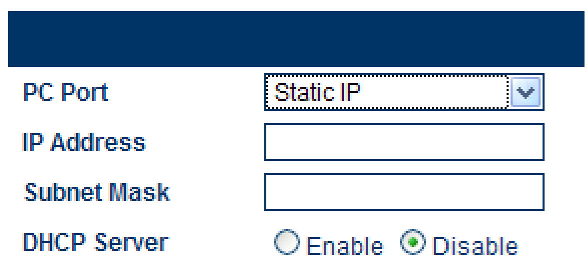


Figure 1-10 PC Port (Fixed IP)

To simplify network IP assignments, enable the DHCP Server for the PC Port. This allows network devices connected Port to obtain network IP and related information from the PC Port. Please consult your network administrator for proper settings of the DHCP Server

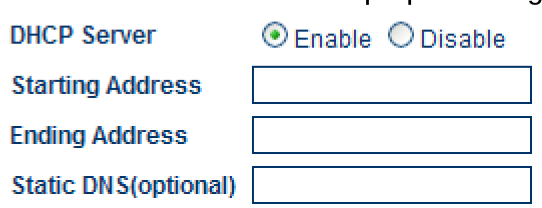


Figure 1-11 PC Port DHCP Host Setting

3.2.3 Call Settings

The Call Settings page configures all related settings for VoIP Service. The EP-8201 is SIP compliant and it supports up to 4-line appearances in two registration modes: **Single Server** and **Multiple Servers**.

1. **Single Server Mode** – This mode allows SIP registrations to only one SIP Server / Proxy; however, it can support up to 4 registrations of different SIP numbers and names (referred as “**Contact**” in the webpage). A backup server option is available and it will be used once registration to the primary server fails. Line 1 to Line 4 keys are predefined for the “**Contact1**”, “**Contact2**”, “**Contact3**”, and “**Contact4**” respectively. This allows the user to specify which identity (Contact: SIP number and name) will be used for the call. The default is to use the contact information for Line1 (“**Contact1**”). Below shows the webpage for **Single Server Mode**.

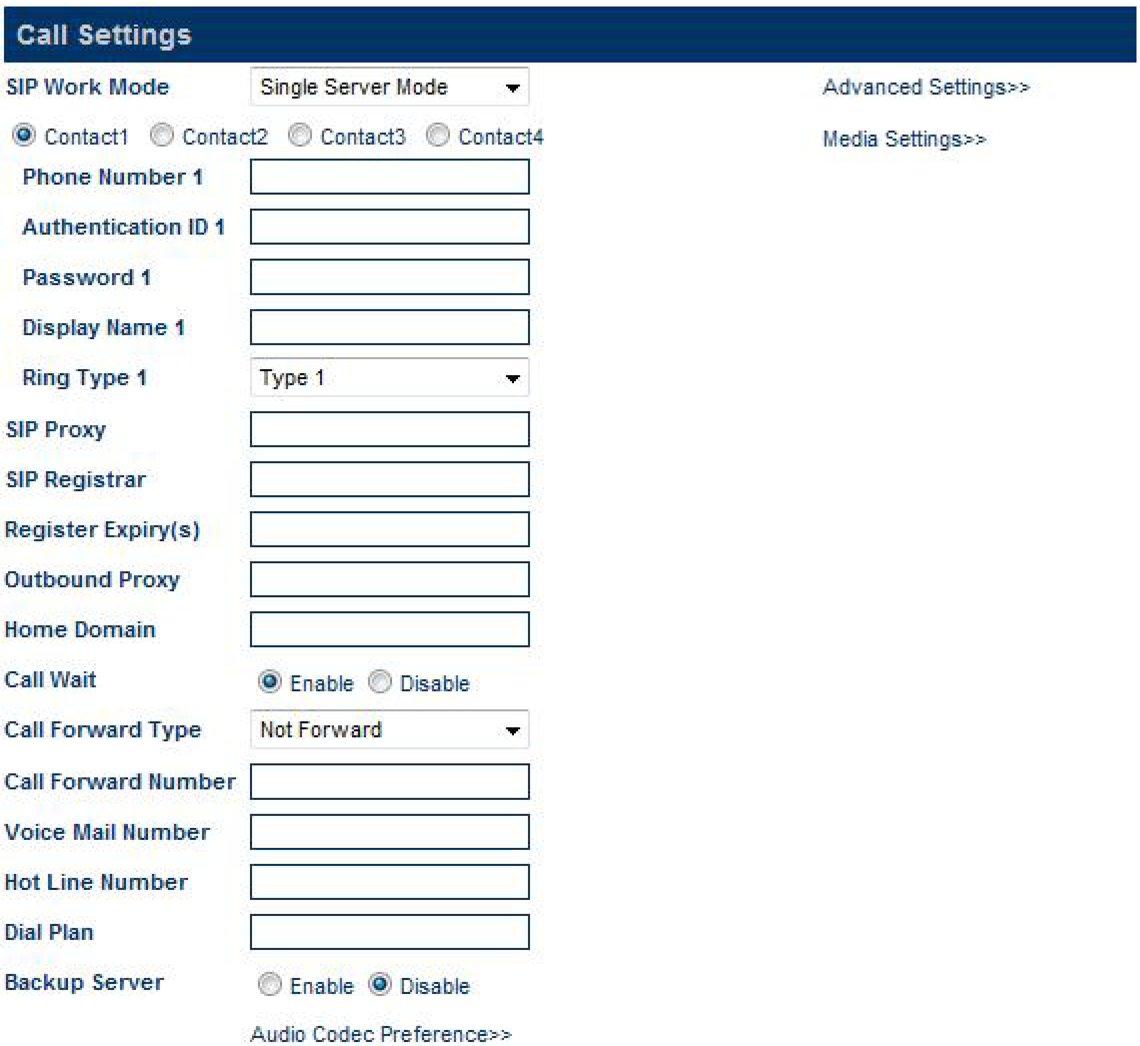


Figure 1-12 Single SIP Server Mode

2. **Multiple Server Mode** – This mode allows up to 4 SIP registrations to up to 4 different SIP servers / proxies as shown below. Each registration is referred as a “**Profile**”. **Line 1** to **Line 4** keys are predefined for **Profile 1** to **4** respectively. If a “**Profile**” is not defined, the corresponding Line key is disabled. If a call is made without pressing a **Line** key, the default line to be used is **Line 1** (**Profile 1**). Below shows the webpage for **Multiple Server Mode**.

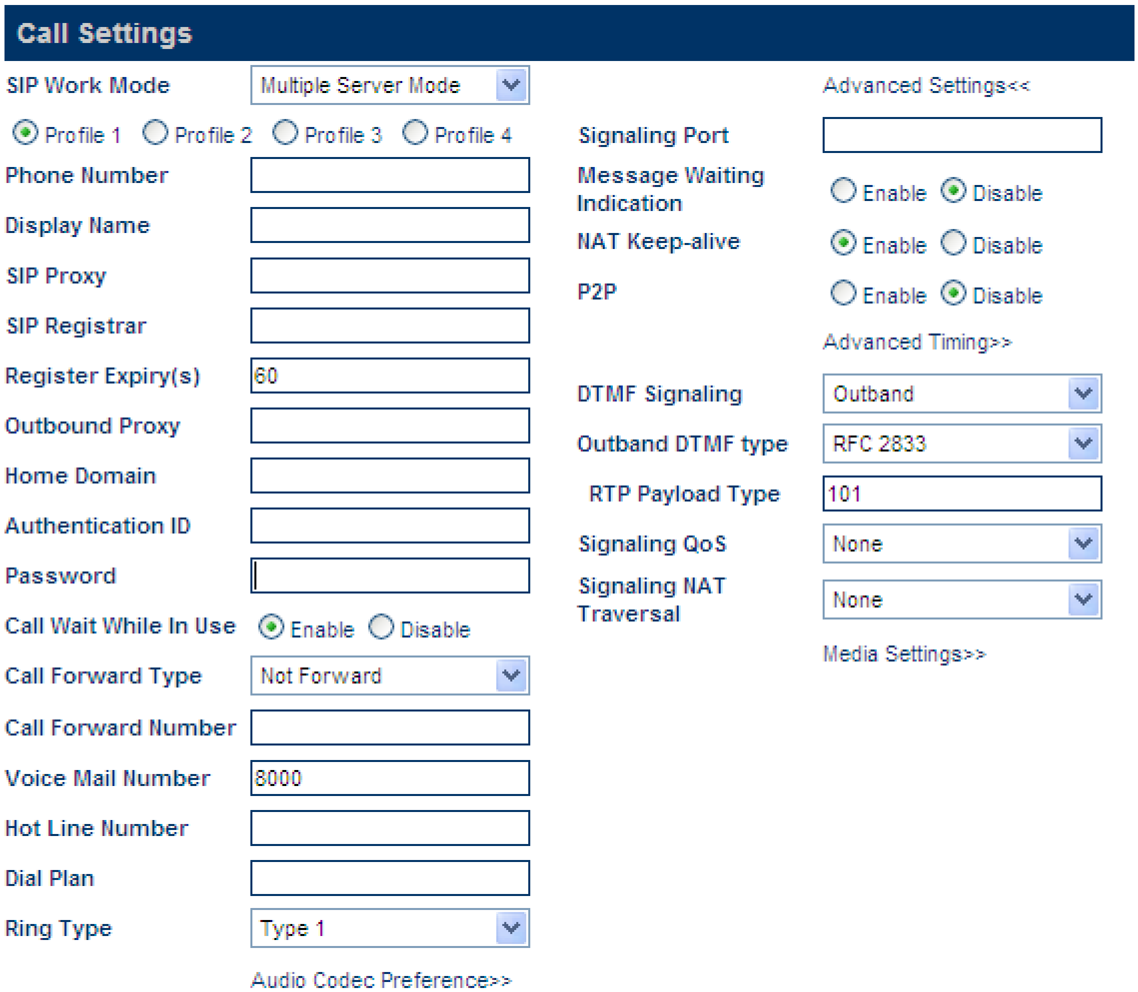


Figure 1-13 Multiple SIP Server Mode

The basic SIP settings are:

1. **Phone Number** – This parameter assigns the phone number used for SIP registration.

2. **Display Name** – This parameter (optional) specifies the Caller name and is transmitted as part of the caller ID.

3. **SIP Proxy** – A SIP Proxy acts as a call manager of all incoming and outgoing calls. Specify the location (IP address / domain name) of the designated SIP Proxy used for SIP service. The standard port used is 5060. To specify a non-standard signaling port, add ":<port number>" to the of the location. For example: If SIP Proxy = yoursippbx.com, the signaling port is the standard port 5060. If SIP Proxy = yoursippbx.com:15060, the signaling port is 15060.

4. **SIP Registrar** – A SIP Registrar maintains a database of all SIP phones registered and their contact information. Specify the location (IP address / domain name) of the designated SIP Registrar. The standard port used is 5060. To specify a non-standard signaling port, add ":<port number>" to the of the location. For example: If SIP Proxy =

yousippbx.com, the signaling port is the standard port 5060. If SIP Proxy = yoursippbx.com:15060, the signaling port is 15060.

5. **Registry Expiry(s)** – This specifies the expiry duration at the SIP Registrar after a successful registration. The range is 60 to 36400 seconds.

6. **Outbound Proxy** – A network node acts as proxy for outbound traffic between a client and a server. Please contact your network administration to determine if this proxy is available or not.

7. **Home Domain** – This field enables the use of home domain name is SIP registration instead of IP address.

8. **Authentication ID** – This field specifies the ID to be used for Authentication during a SIP registration.

9. **Password** –This field specifies the password used for Authentication during a SIP registration.

10. **Call Forward Type** – This defines the Call Forward condition and the available options are:
    a) **Not Forward** – Call forward is disabled.
    b) **Unconditional Forward** – Call is always forwarded.
    c) **Busy Forward** – Call is forwarded when the line is in use / engaged.
    d) **No Answer Forward** – Call is forwarded when it is not answered.
    e) **Busy or No Answer Forward** – Call is forwarded when the line is in use or not answered.

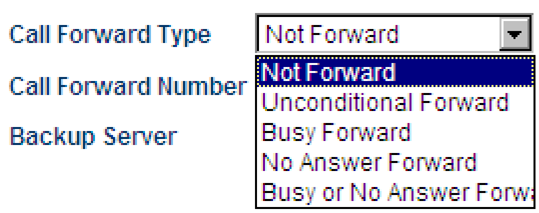


Figure 1-14 Call Forward Setting

11. **Forward Number** – This defines the number to be used for Call Forward.

12. **Voice Mail Number –** This defines the number for voice mail server.

13. **Hotline Number** – When this feature is enabled, the Hot Line Number defined will be dialed out automatically whenever the phone is off hook. This feature is only available for VoIP calls; **Line 1 Default** is set to **VoIP**

14. **Dial Plan (Digit Map)** – Dial Plan defines how a number (VoIP) is processed when the device receives it. This field is located in the Calling Setting Window and it is available for both H.323 Phone and SIP Phone. The Dial Plan is very flexible and can be configured for a wide range of dialing applications.

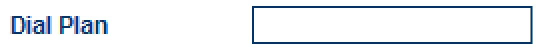


Figure 1-15 Dial Plan

The basic syntax is “<**event**>:<**action**>|<**event**>:<**action**>|…”, where

<**event**> defines the event to be matched. A event consists of a sequence of digits. If a specific digit has a limited range, use the syntax [A-B] where A and B are both digit (0 to 9) and B is greater than A. The length of the input number can be limited by using “X” to represent each unknown digit. If this field is omitted, it means any event.

<**action**> defines the action to be taken on the number received and it consists of “–“ (minus), “+” (plus), and digits. “-“ followed by digits means to remove the digits from the beginning of the number entered. “+” followed by digits means to add the digits in front of the number entered.

“**|**” means or and the order of priority is from left to right.

Note: For practical use, it should not be possible to reach the maximum length of the Dial Plan string.

Examples:

1. Dial Plan = “010:-010” means that the number dialed out will have the first 3 digits ”010” removed when a number with the first digits as “010” is entered.
   a) Number entered = “01082121234”, actual number dialed = “82121234”.
   b) Number entered = “82121234”, actual number dialed = “82121234”.

2. Dial Plan = “1:+00” means that the number dialed out will have the “00” added in front of the number entered when a number with the first digit as “1” is entered,.
   a) Number entered = “1082121234”, actual number dialed = “00182121234”.
   b) Number entered = “82121234”, actual number dialed = “82121234”.

3. Dial Plan = “001:-001+1751” means that the number dialed out will the first 3 digits “001” changed to “1751” when a number with the first digits as “001” is entered.
   a) Number entered = “00182121234”, actual number dialed = “175282121234”.
   b) Number entered = “82121234”, actual number dialed = “82121234”.

4. Dial Plan = “XXXX:” means that the input number is limited to 4-digit long and will be dialed out immediately when the fourth digit is entered.

5. Dial Plan = “13XXXXXXXXX:+0” means that the input number is restricted to 11-digit long and the first two digits must be “13”. When this condition is matched, the number dialed out will have a leading “0” added.
   a) Number entered = “13901234567”, actual number dialed = “013901234567”.
   b) Number entered = “12801234567”, actual number dialed = “12801234567”.

6. Dial Plan = “13[6-9]XXXXXXXX:+0” means that the input number is restricted to 11-digit long and the first two digits must be “13” and the third digit can be 6, 7, 8,or 9. When this condition is matched, the number dialed out will have a leading “0” added.
   a) Number entered = “13901234567”, actual number dialed = “013971234567”.
   b) Number entered = “13001234567”, actual number dialed = “13001234567”.

Please note that the above samples are simple and intended to show the meaning of various rules. They may not have any practical meaning. A combination of these rules (joined with the symbol “|”) can be realized for a much more complicated dialing application.

15. **Backup Server (Single Server Mode only)** – The backup option provides settings for a SIP backup server. Once the designated SIP Proxy and/ SIP Registrar fail, the backups will be used automatically.

Advanced Settings

More settings are available under the ***Advanced Settings*** tab. Depending on your network requirements, please consult your network administrator for the correct configuration.

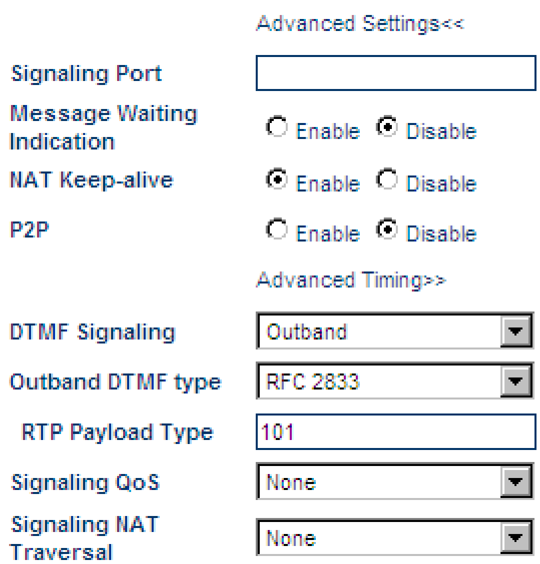


Figure 1-16 Advanced Setting

1. **Signaling Port** – This Port is used to convey signaling message with the SIP Proxy. The standard port number is 5060.

2. **NAT Keep-alive** – When enabled, a dummy packet I sent to the local firewall / router in order to keep the ports opened for VoIP service.

3. **P2P** – This enables Peer-to-Peer calls.

4. **Virtual Ringback** – This enables a ringback tone to be generated whenever a call is made.

5. **DTMF Signaling** – This parameter sets the method of sending DTMF signals. **Inband** measns that the DTMF signal is sent as an analog signal via the voice channel. **Outband** means that the DTMF signal is sent as DTMF command via the data channel. Both **RFC2833** and **SIP INFO** methods are supported. For **RFC2833,** a DTMF payload type is required and the default type is set to 101.

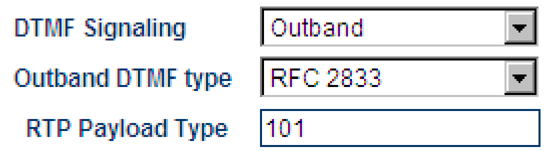


Figure 1-17 DTMF Signaling Setting

6. **Signaling QoS** – This parameter sets the QoS mode for VoIP Signaling for better response time and more reliable VoIP Call signaling. Both IP TOS and Diffserv modes are supported. Please check with your network administrator or ISP for the correct setting.

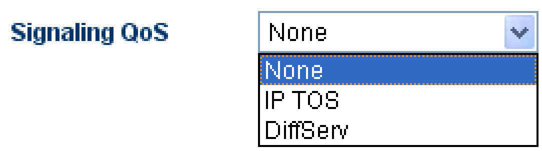


Figure 1-18 Signaling Q0S

7. **Signaling Encryption** – Three types of encryption are supported and they can be enabled individually. These are non-standard encryption for signaling. Please make sure that your SIP Service Provider can support the encryption(s) enabled.

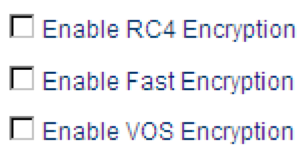


Figure 1-19 Signaling Encryption

a) **RC4 Encryption** – RC4 Encryption Key is required when it is enabled.
b) **Fast Encryption** –
c) **VOS Encryption** – This encryption is used mainly in China.

8. **Signaling NAT Traversal** – NAT Traversal is an algorithm designed to solve a common problem in TCP/IP networking in establishing connections between hosts in private TCP/IP networks that use NAT devices. This parameter only sets the NAT Traversal mode for VoIP signaling. The 2 methods supported are **STUN(RFC3489)** and **Relay Proxy**. A STUN Server is required for **STUN(RFC3489)**.

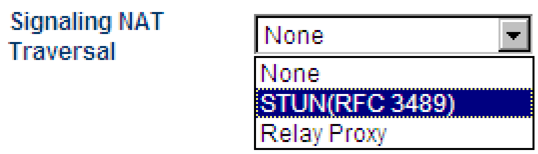


Figure 1-20 Signaling NAT Traversal

Relay Proxy mode is a proprietary NAT protocol and it requires the use of our Relay Proxy Server. All VoIP signaling packets are encapsulated (encrypted for more secured transmission if enabled) and transmitted via another port/channel.

***Note**: For Service providers, RELAY Proxy software is available at no charge. Please contact your supplier for support. For end user, please contact your service provider to see if this feature is available.*

Media Settings

Once a VoIP call is established, the Media channel is used for voice transmission. The settings listed below configure the performance and operation of the Media channel.

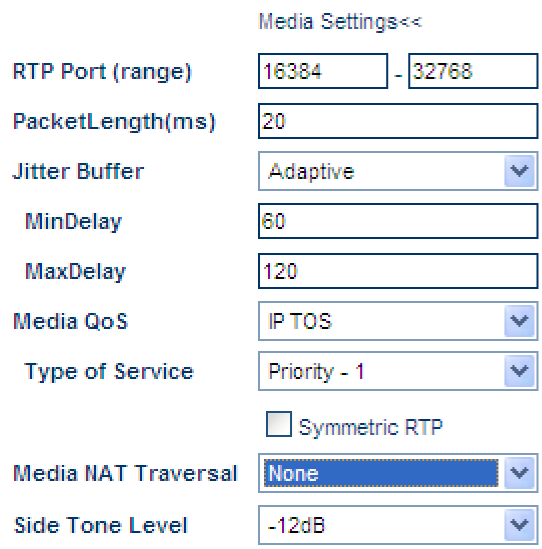


Figure 1-21 Media Settings

1. **RTP Port (range)** – Audio stream is transmitted via Real Time Protocol (RTP) and at least 4 ports are used per voice channel. The default port range is 16384 – 32768. Specify the port range depending on your network environment if needed.

2. **Packet length (ms)** – This specify the length of a voice packet. The default packet length is 20 ms.

3. **Jitter Buffer Mode** –Three jitter modes are available. The **Fixed Mode**, which is the default mode, is a simple first in first out mode, with a fixed jitter buffer delay. By definition the jitter buffer depth is twice the jitter buffer delay. The **Sequential Mode** is also a fixed jitter buffer delay mode, but in this mode the jitter buffer function looks at the packet timestamp for dropped or out of sequence packet problems. The data packets are sorted based on the

packet timestamp. The **Adaptive Mode** optimizes the size of the jitter buffer delay and depth in response to network conditions, in addition to the sequential mode.

4. **Media QoS** – QoS is also available for Media packets to improve voice quality. This is rather significant in a network environment with large amount of data traffics. Both **IP TOS** and **DiffServ** methods are supported.

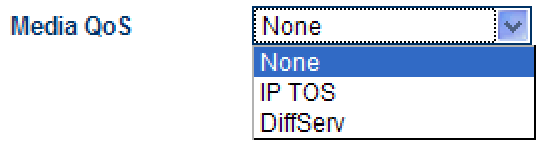


Figure 1-22 Media QoS

5. **Encryption** – For secure voice transmission, RC4 Encryption can be enabled for media channel. Please make sure your service provider can support this encryption method before enabling this feature.

6. **Symmetric RTP** – Enable the media channel to use symmetric RTP ports. Some network environment demand the use of Symmetric RTP.
7. **Media NAT Traversal** – NAT Traversal can be set independently for Media packets. This gives a more flexible setting for various network environment. Three modes are supported: **STUN(RFC 3489)**, **Port-forward/DMZ**, and **Relay Proxy**.

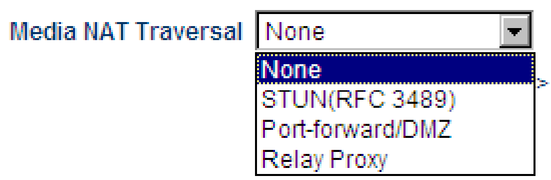


Figure 1-23 Media NAT Traversal

Relay Proxy mode is a proprietary NAT protocol and it requires the use of our Relay Proxy Server. All VoIP signaling packets are encapsulated (encrypted as well if enabled) and transmitted via another port/channel. Three relay modes of operation are supported.

Mode 1: Use UDP packets and encryption.

Mode 2: Use UDP packets and encryption; use single UDP port.

Mode 3: Use TCP packets and encryption; Use single TCP port;

The mode 2 and mode 3 are the passive and the port use is assigned by the RELAY SERVER.

Note: For Service providers, RELAY Proxy software is available at no charge. Please contact your supplier for support. For end user, please contact your service provider to see if this feature is available.

8. **Audio Codec Preference** – The table below list the voice codec priorities in descending order. Each voice codec can be enabled (place a check mark in the check box) or disabled individually. Select the voice code and then click on the UP or DOWN button to move the order on the list.

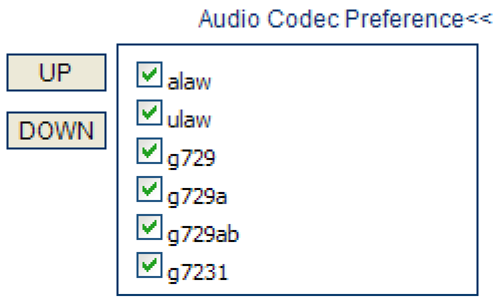


Figure 1-24 Codec Preference

3.2.4 Phone Settings

The Phone Settings page configures the phone related operations. They are described in details below.

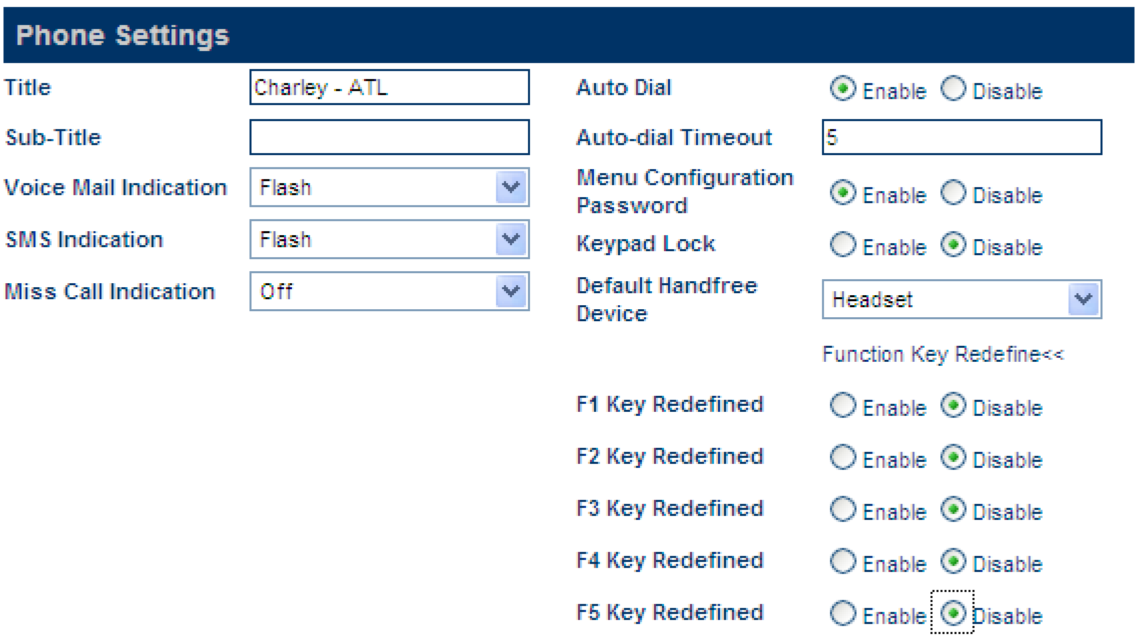


Figure 1-25 Phone Settings

1. **Title on LCD** – The text entered here is shown on the LCD as a default message when the phone is idle.

2. **Auto Dial** – This field enables the number entered on the LCD to be dialed out automatically once the **Auto-Dial Timeout** expires.

3. **Auto-dial Timeout** – This field specifies the timeout duration (in second) for automatically dialing out the number entered.

4. **Menu Configuration Password** – This field specifies if a login ID / password is required to access the **Device CFG** submenu in the **Phone Menu**.

5. **Voice Mail Indication** – This field sets the LED on the top right hand corner to be used to flash or illuminate when a Voice Mail is received.

6. **SMS Indication** – This field sets the LED on the top right hand corner to be used to flash or illuminate when a SMS message is received.

7. **Missed Call LED Indication** – This field sets the LED on the top right hand corner to be used to flash or illuminate when a miss call occurs.

8. **Keypad Lock** – This parameter enables or disables the keypad lock feature. When it is enabled, an unlock password is required to be entered. To activate keypad lock, dial ***99**. To deactivate keypad lock, dial ***99** and then enter the **Unlock Password** + "**OK**".

9. **Default Handsfree Device** – This parameter defines whether the **Speakerphone** or the Headset is for handsfree operation. Press the Speaker key to activate the handsfree operation.

10. Function Keys Redefine – There are 5 function keys (R1 to R5) that are user programmable. Each key can be enabled or disabled individually and can be redefined in two modes: **Fixed** and **Manual**.

In **Fixed** mode, five predefined functions are available for selection as shown below. They are **Phone Book**, **Do Not Disturb**, **Show Pending Call List**, **Call Forward**, and **Speed Dial**.

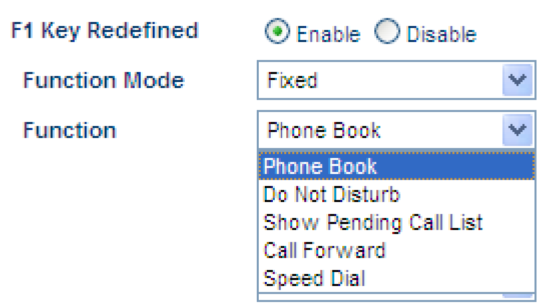


Figure 1-26 Programmable Keys – Predefined Functions

In **Manual** mode, each Speed Dial and DTMF Speed Dial

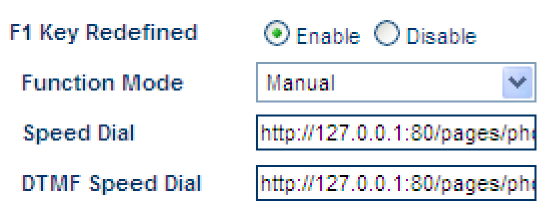


Figure 1-27 Programmable Keys -

3.2.5 Save Changes

When all changes have been made, click on the **Save Changes** tab to save all settings to the Flash memory.

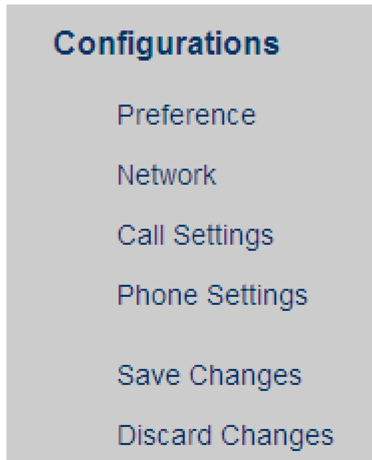


Figure 1-28 Configurations Menu

The message window below is displayed when the saving is completed.

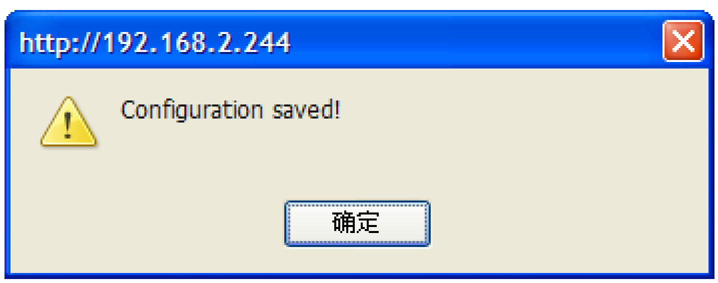


Figure 1-29 Popup Message for Configuration Saved

3.2.6 Discard Changes

Click on the **Discard Changes** tab to ignore all changes made.

4.3 Phone Book

The EP-8201 has a built-in phone book with a maximum of 250 entries. Each entry contains a **Number** field and a **Name** field. The maximum character for each field is 32 characters and there is no restriction on the characters entered. The **Number** field is used to make the VoIP

call. The **Name** field is used for identification. It can be used for searching a phone book entrie (will be supported in future version).

Click on the **Phone Book** selection in the left hand menu column accesses the Phone Book Page shown below directly. All existing entries are displayed in ascending alphabetical order of the Name field.

No.	Name	Number		
1	Craig0	6900	Edit	Del
2	Craig1	6901	Edit	Del
3	Craig10	6910	Edit	Del
4	Craig100	6800	Edit	Del
5	Craig101	6801	Edit	Del
6	Craig102	6802	Edit	Del
7	Craig103	6803	Edit	Del
8	Craig104	6804	Edit	Del
9	Craig105	6805	Edit	Del
10	Craig106	6806	Edit	Del
11	Craig107	6807	Edit	Del
12	Craig108	6808	Edit	Del
13	Craig109	6809	Edit	Del
14	Craig11	6911	Edit	Del
15	Craig110	6810	Edit	Del
16	Craig111	6811	Edit	Del

Figure 1-30 Phone Book Page

3.3.1 Edit a Phone Book Entry

To edit a phone book entry, just click on the Edit icon on the right hand side to access the Edit

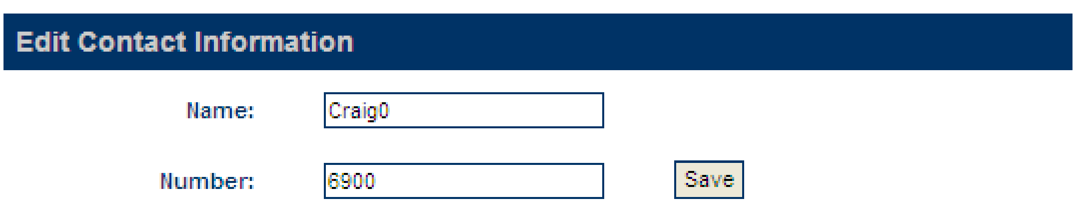


Figure 1-31 Edit a Phone Book Entry

3.3.2 Delete a Phone Book Entry

To delete a phone book entry, just click on the Del icon on the right hand side.

3.3.3 Add a Phone Book Entry

To add a phone book entry, just click on the Add selection on the left hand menu column to access the Add New Contact page as shown below.

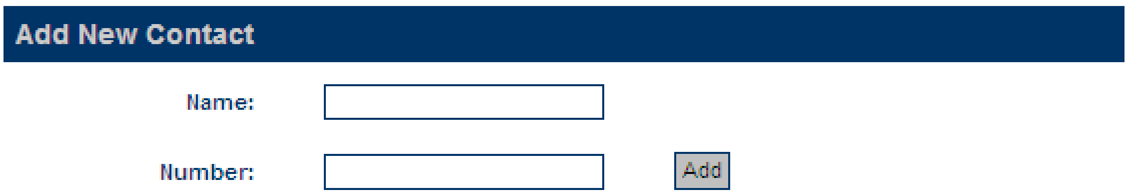


Figure 1-32 Add a Phone Book Entry

The **Number** field must be entered properly in order for the dialing of this entry to be successful. The **Name** field is optional and it is not used for actual dialing. Click Add **to add the entry to the Phone Book.**

3.3.4 Backup / Restore Phone Book

This function backups or restores the phone configuration.

Click on the Save button to save the current configuration as **config.dat** which is placed on the desktop. The **Password** field is optional. If a password is entered, the file **config.dat** is encrypted and password protected.

To restore a configuration file, click on Browse button to locate the file and enter the password, if required. Then click on Restore to restore the phone configuration.

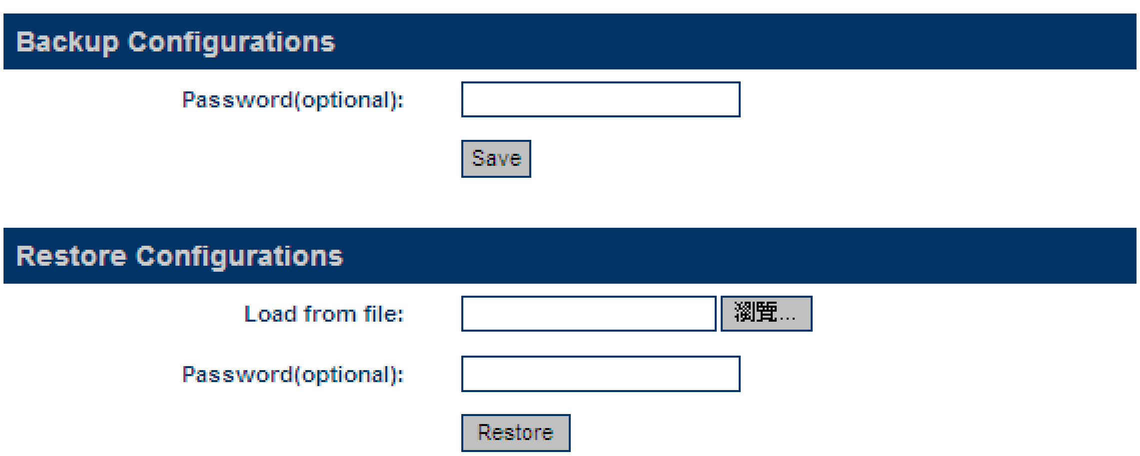


Figure 1-33 Phone Book Backup / Restore

3.3.5 Auto Update

The Auto Update feature allows the phone book to be updated automatically and periodically via a http / ftp / tftp server. Only xml file format is supported.

Click on the Auto Update selection in the left hand menu column to access the page below.

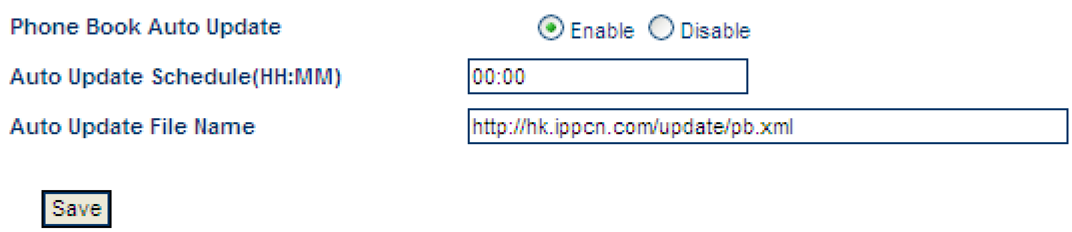


Figure 1-34 Phone Book Auto Update

Select Enable to activate this feature. Enter the Auto Update Schedule in the 24 hour format HH:MM and the Auto Update File Name including the path as shown above. Once they are completed, click on Save to save the settings. When the schedule time is reached, the phone will update the phone book with the file specified. If the specified file cannot be read successfully, an error message is displayed on the LCD. Pressing any key will clear this message.

Note: Please note that the amount of time required to perform the update depends on the number of entries in the xml documents since it takes time to write the phone entries to the Flash memory. During the Auto Update, the phone operation could be affected. It is recommended to set the Auto Update schedule to a time when the phone is rarely used (say the middle of the night).

4.4 Tools

The **Tools** section is intended to offer the following functions: Online Upgrade, Change Password, Backup/Restore Configurations, Reset Config, and Reboot as shown below.

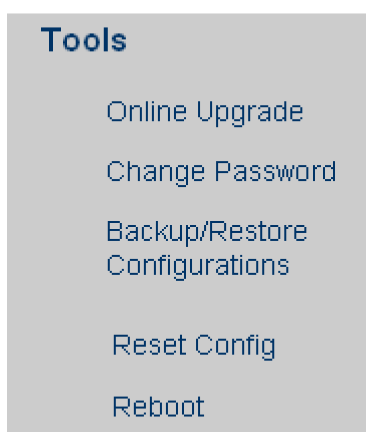


Figure 1-35 Tools Menu

3.4.1 Tools

Click on the **Online Upgrade** tab to perform manual firmware upgrade. Enter the upgrade address as shown below. Please contact your service provider to determine if there is a new firmware available.

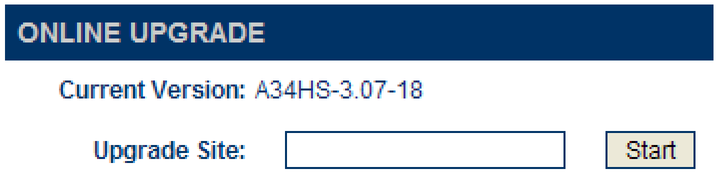


Figure 3-2 Firmware Upgrade

WARNING: Once the upgrade starts, a message window is displayed to show the upgrade status.

DO NOT TURN OFF THE POWER WHILE THE FIRMWARE UPGRADE IS IN PROCESS!

3.4.2 Change Password

EP-8201 supports two login levels to the built-in webpage. The User level is intended for general user and is restricted from accessing the **Call Settings** page and **Reset Configuration** function. In this level, only the password for the user level can be changed. The default password for the user level (login ID = user) is “1234”.

The Administrator level allows full accessing to the EP-8201 configurations. In this level, the password for both levels can be change. The default password for the administrator level (login ID = admin) is “admin”.

It is important to keep a record the new password(s) entered. Please contact your local support if the password is forgotten.

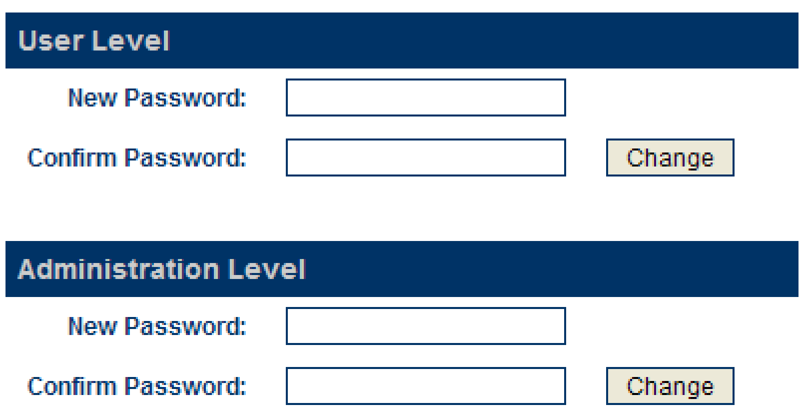


Figure 3-3 Change Password

3.4.3 Backup / Restore Configurations

This function backups or restores the phone configuration.

Click on the Save button to save the current configuration as **config.dat** which is placed on the desktop. The **Password** field is optional. If a password is entered, the file **config.dat** is encrypted and password protected.

To restore a configuration file, click on Browse button to locate the file and enter the password, if required. Then click on Restore to restore the phone configuration.

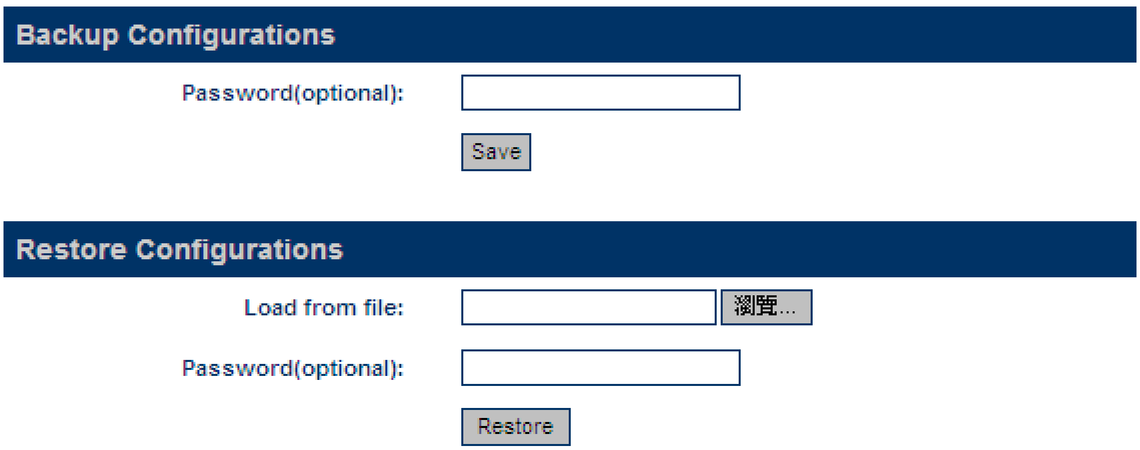


Figure 3-4 Configuration Backup / Restore

3.4.4 Reset Configuration

This function can only be accessed in administrator login level. Click on the **Reset Configuration** tab to initiate the reset process. A message windows pops up to ask for confirmation. Click "Yes" to reset all configurations back factory defaults. Click "No" to cancel. Once the reset process is completed, the device reboots itself.

3.4.5 Reboot

Click on the **Reboot** tab to reboot the device. This will take couple minutes.

4.5 Gain Settings

This **Gain Settings** page is hidden and is only intended for advanced user who is technically capable. The block diagram below shows the various gain stages in the phone.

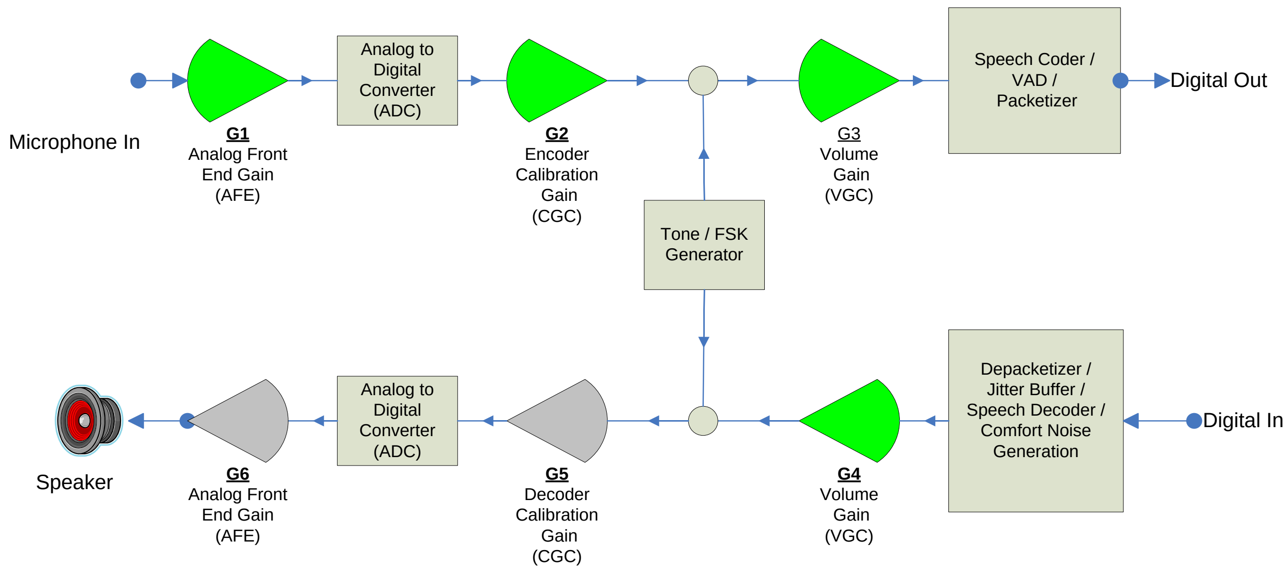


Figure 1-36 Block Diagram for Internal Gain Settings

The blocks highlighted in green are available for programming in the URL below.

[http://xxx.xxx.xxx.xxx//en_US/**gain.html**](http://xxx.xxx.xxx.xxx//en_US/gain.html)

Figure 1-37 GUI for Internal Gain Settings

The first 3 rows in the gain settings page shows the 3 modes of operation: Handset, Handsfree, and Headset. Each mode uses the same configuration as shown in the block diagram above. The left hand column (first 3 rows) shows the mode of operation. The various gain settings are described as follows:

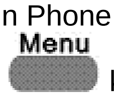
1. Microphone AFE refers to the Analog Front End Gain block (**G1**).

2. Microphone VGC refers to the Volume Gain (<u>**G3**</u>). This gain should be changed with caution since it affects the tone level as well. If the tone generated is a DTMF tone, changing this gain may affect the detection of the DTMF tone by the outside network.
3. Speaker refers to the Volume Gain (<u>**G4**</u>).
4. Microphone CGC on the 4th row in the Gain Settings page refers to the Encoder Calibration Gain CGC (<u>**G2**</u>). Changing this gain does not affect the signals from the Tone / FSK generator.
5. Speaker Levels refer to the relative levels in the volume settings. There are a total of 6 volume settings from 1 to 6. The relative levels are listed in ascending order. When the volume setting is set to 1, the Decoder Calibration Gain CGC (<u>**G5**</u>) is set to its preset value (cannot be changed) + the relative level specified for volume setting 1. If the relative level is blank, <u>**G5**</u> is set to its minimum value. This means that the speaker is effectively muted.
6. The Analog Front End Gain (<u>**G6**</u>) is factory preset and cannot be changed.

4

Phone Menu

The built-in Phone Menu allows configuration to the phone manually via the phone keypad.

Press the 

key to activate the Phone Menu. There are five main categories as shown on the LCD:

MAIN MENU:

1. CALL HISTORY
2. PHONE BOOK
3. MESSAGE CENTER
4. SYSTEM TOOLS
5. DEVICE CONFIG

Press  or  to scroll the menu selections and the press 

to select (or press the corresponding menu item number).

Please refer to the Appendix C for the complete Phone Menu for Multiple Server Mode.

4.1 Call History

The **CALL HISTORY** allows the user to view, edit or dial the caller information. For Single Server Mode, the call history is organized with respect to each SIP Number programmed ("Contact x"). For Multiple Server Mode, the call history is organized with respect to each profile defined. Therefore, the user must first select the corresponding contact or profile desired first.

CALL HISTORY:	CALL HISTROY
1. CONTACT 1	1. PROFILE 1
2. CONTACT 2	2. PROFILE 2
3. CONTACT 3	3. PROFILE 3
4. CONTACT 4	4. PROFILE 4

Once this is done, the next menu is to choose the call history category (ANSWERED, MISSED, DIALED) or to delete all call history.

CONTACT x:	PROFILE x:
1. ANSWERED CALLS	1. ANSWERED CALLS
2. MISSED CALLS	2. MISSED CALLS
3. DIALED CALLS	3. DIALED CALLS
4. DELETE ALL	4. DELETE ALL

The call entries of the selected category are then displayed. Press  or  to scroll the call entries and then press  to select the highlighted entry. The following menu is then displayed to prompt for further action to be performed on the selected entry. Select 1 to dial the entry, 2 to save the entry to the phone book, 3 to delete the entry, and 4 to view more detailed information of the entry.

XXXXXXX:

1. DIAL
2. SAVE
3. DELETE
4. DETAILS

4.2 Phone Book

The **PHONE BOOK** menu allows the user to manage the Phone Book/Speed Dial Memory or to dial an entry in the Phone Book.

PHONE BOOK:

1. VIEW
2. ADD NEW
3. SPEED DIAL

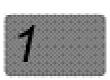
1. Select VIEW to manage or dial the phone book entry. Once an entry is selected, the user can then choose from the menu to DIAL, EDIT, or DELETE the entry.
2. Select ADD NEW to add a new phone book entry. The selection allows the user to enter a new phone book entry (Name and Number).
3. Select SPEED DIAL to manage the Speed Dial Memory. The 10 Speed Dial entries (empty or not) are displayed. Choose the Speed Dial entry / location and a new menu for CHANGE or DELETE is displayed. Choose CHANGE to assign a phone book entry to this location. Choose DELETE to clear this location.

4.3 Message Center

The **MESSAGE CENTER** allows the user to access SMS and Voice Mail.

MESSAGE CENTER:

1. SMS
2. VOICE MAIL

1. Press 

to select SMS. The User is then prompted to select the **Profile** or **Contact** desired. The user is then ready to access SMS via the menu shown below. Select INBOX to read the existing messages or WRITE SMS to write and then send a SMS.

SMS:

1. INBOX
2. WRITE SMS

2. Press 

and then the **Profil**e or **Contact** to access the Voice Mail directly. Please note that the voice mail number must be predefined in the **Call Settings** Page for the corresponding **Profile** or **Contact**.

4.4 System Tools

The **SYSTEM TOOLS** menu has 9 selections for general phone management and information viewing.

MAIN MENU:

1. PHONE STATUS
2. ONLINE UPGRADE
3. SYSTEM VERSION
4. RESET CONFIG
5. RING TYPE
6. RING VOLUME
7. CALL FORWARD
8. DEF. HANDFREE DEV.
9. REBOOT

1. **PHONE STATUS** – Select this to view the information assigned to the LAN port, PC port, PHONE NUMBER, and HW INFO (hardware information). LAN port and PC port information includes the current status, IP address, gateway address, and Netmask. PHONE NUMBER includes the sip number assigned to each line.

2. **ONLINE UPGRADE** – Select this to initiated online upgrade. **User Name** and **Password** are required before entering the upgrade site. Please consult your local support for the latest firmware version.

3. **SYSTEM VERSION** – Select this to view the current version of both software and hardware.
4. **RESET CONFIG** – Select this to reset he phone to factory default settings. **User Name** and **Password** are required.
5. **RING TYPE** – Select this to assign the ring type of each line.
6. **RING VOLUME** – Select this to set the ringing volume of all incoming calls.
7. **CALL FORWARD** – Select this to programs the Call Forward settings for all lines.
8. **DEF. HANDFREE DEV.** – Select this to set the default device (speakerphone or headset) used for handsfree operation.
9. **REBOOT** – Select this to reboot the phone. No confirmation is required.

4.5 Device Config

The DEVICE CONFIG menu configures the phone preference, network settings, and DHCP service. **User Name** and **Password** are required in order to access this menu.

DEVICE CONFIG:

1. PREFERENCE
2. NETWORK CONFIG
3. DHCP SERVICE

1. **PREFERENCE** – Select this to configure the following settins: Language, Time Zone, Date Time Server, and AUTO Configuration server.
2. **NETWORK CONFIG** – Select this to configure the DNS, Gateway, PC port, LAN port, and VLAN.
3. **DHCP SERVICE** – Select this to enable or disable DHCP host service for the PC port.

5

Phone Operation

5.1 Making a Call

1. Pick up handset.
2. Dial a phone number.
3. Press OK or wait for Auto Dial timeout (# key as well if enabled). Depending on the configuration mode, the default **Contact1** or **Profile1** will be used to make the call. (See notes below)

5.2 Making a Hands-Free Call

1. Press Speaker**.**
2. Dial phone number.
3. Press OK or wait for Auto Dial timeout (# key as well if enabled). Depending on the configuration mode, the default **Contact1** or **Profile1** will be used to make the call. (See notes below)

Notes:

a. Instead of pressing OK, press one of the line keys (L1, L2, L3, or L4**)** to select the **Contact** or **Profile** to be used for the call.

b. An alternative way to make a call is to press a line key (L1, L2, L3, or L4**)** to select the appropriate line first before dialing a phone number.

5.3 Answering an Incoming Call

There are two ways to answer an incoming call:

1. Pick up the handset to answer the call normally.
2. Press the Speaker button to answer in speakerphone mode.

5.4 Dialing from the Phonebook

1. Press Menu
2. Choose **PHONE BOOK** (Press 2ABC)
3. Choose the Profile desired (for Multiple Server Mode only)
3. Choose **VIEW** (Press 1)
4. Press ▲ (**UP**) or ▼ (**Down**) to view the Phone Book
5. Press OK to select the highlighted entry
6. Select **DIAL** to dial out the number (Press 1)

5.5 Viewing / Dialing from the Call History

1A. Press ▲ to view the Missed Call List while on hook / idle

1B. Press ▼ to view the Answered Call List while on hook / idle

1C. Press OK to view the Dialed Call List while on hook / idle

2. Choose the Profile desired (for Multiple Server Mode only)
3. Press ▲ or ▼ to view the selected Call List
4. Press OK to dial out the highlighted entry

5.6 Redialing the Last Number

1. Pick up handset

2. Press L1, L2, L3, or L4 to select a line

3. Press Redial to dial out the last number dialed immediately

Or

1. Press Redial

2. Press ▲ or ▼ select/high light the last number dialed from L1 to L4.

3. Press OK to dial the number selected. The phone goes into Speakerphone mode automatically and there is no need to select the line.

4. Pick up the handset to talk directly (Speakerphone mode turns off automatically).

5.7 Speed Dial

The Speed Dial function is available if one of the function keys (R1, R2, R3, R4, or R5) is defined as **Speed Dial**. Press this Speed Dial key to activate the Speed Dial Entries to be shown on the LCD. Select the desired entry and then press OK to dial it out.

5.8 Putting / Releasing a Call on Hold

To put a call on hold:

1. Press Hold button

To release a call on hold:

1. Press Hold button

5.9 Transferring a Call

To transfer a call to another extension:

1. Press Transfer **Transfer** button

2. Dial phone number

3. Press OK

4A. Hang up for unattended transfer.

4B. Wait for the call to be answered and then hang up for attended transfer.

5.10 Answering a Call Waiting Call

When you are talking on the phone and another call comes in on your phone extension, a short tone sounds in your handset and the LCD displays an incoming call message.

To answer a Call Waiting Call:

1. Press the corresponding line key with an illuminated LED to put the current call on hold and answer the Call Waiting Call.
2. Press the line key (Line) again to switch between the two calls.

5.11 Adjusting the Ringing Volume

1. Press Menu.
2. Select SYSTEM TOOLS.
3. Select RING VOLUME.
4. Press or to increase or decrease the ring volume as shown on the LCD.

5.12 Adjusting the Handset Receiver Volume

1. Pick up the handset
2. Press or to increase or decrease the handset receiver volume as shown on the LCD.

5.13 Adjusting the Speaker Volume

1. Press the **Speaker** button.
2. Press or to increase or decrease the speaker volume as shown on the LCD.

5.14 Adjusting the LCD Contrast

1. While the phone is idle, press  or  to increase or decrease the LCD Contrast level as shown on the LCD.

5.15 Resetting Phone Configuration

To reset the SIP Phone to factory configuration:

1. Press .
2. Select SYSTEM TOOLS (Press ).
3. Select RESET CONFIG (Press ).
4. Press .
5. Enter the Username (default is admin) and Password (default is dbl#admin).

This will reset the entire phone configuration back to factory default settings.

Appendix A: Dial Plan

Dial Plan defines how a number (VoIP) is processed when HT-812P receives it. This field is located in the Calling Setting Window and it is available for both H.323 Phone and SIP Phone. The Dial Plan is very flexible and can be configured for a wide range of dialing applications.

The basic syntax is "<**event**>:<**action**>|<**event**>:<**action**>|…", where

<**event**> defines the event to be matched. A event consists of a sequence of digits. If a specific digit has a limited range, use the syntax [A-B] where A and B are both digit (0 to 9) and B is greater than A. The length of the input number can be limited by using "X" to represent each unknown digit. If this field is omitted, it means any event.

<**action**> defines the action to be taken on the number received and it consists of "–" (minus), "+" (plus), and digits. "-" followed by digits means to remove the digits from the beginning of the number entered. "+" followed by digits means to add the digits in front of the number entered.

"|" means or and the order of priority is from left to right.

Note: For practical use, it should not be possible to reach the maximum length of the Dial Plan string.

Examples:

a. Dial Plan = "010:-010" means that the number dialed out will have the first 3 digits "010" removed when a number with the first digits as "010" is entered.
   i. Number entered = "01082121234", actual number dialed = "82121234".
   ii. Number entered = "82121234", actual number dialed = "82121234".

b. Dial Plan = "1:+00" means that the number dialed out will have the "00" added in front of the number entered when a number with the first digit as "1" is entered,.
   i. Number entered = "1082121234", actual number dialed = "00182121234".
   ii. Number entered = "82121234", actual number dialed = "82121234".

c. Dial Plan = "001:-001+1751" means that the number dialed out will the first 3 digits "001" changed to "1751" when a number with the first digits as "001" is entered.
   i. Number entered = "00182121234", actual number dialed = "175282121234".
   ii. Number entered = "82121234", actual number dialed = "82121234".

d. Dial Plan = "XXXX:" means that the input number is limited to 4-digit long and will be dialed out immediately when the fourth digit is entered.

e. Dial Plan = "13XXXXXXXXX:+0" means that the input number is restricted to 11-digit long and the first two digits must be "13". When this condition is matched, the number dialed out will have a leading "0" added.
   i. Number entered = "13901234567", actual number dialed = "013901234567".
   ii. Number entered = "12801234567", actual number dialed = "12801234567".

f. Dial Plan = "13[6-9]XXXXXXXX:+0" means that the input number is restricted to 11-digit

long and the first two digits must be "13" and the third digit can be 6, 7, 8,or 9. When this condition is matched, the number dialed out will have a leading "0" added.

i. Number entered = "13901234567", actual number dialed = "013971234567".
ii. Number entered = "13001234567", actual number dialed = "13001234567".

Please note that the above samples are simple and intended to show the meaning of various rules. They may not have any practical meaning. A combination of these rules (joined with the symbol "|") can be realized for a much more complicated dialing application.

Appendix B: Keypad Encoding

When entering phone book entries, you must use the Phone keypad to key in alphanumeric characters and special characters. The Phone uses the key encoding system similar to the one found in cellular phones.

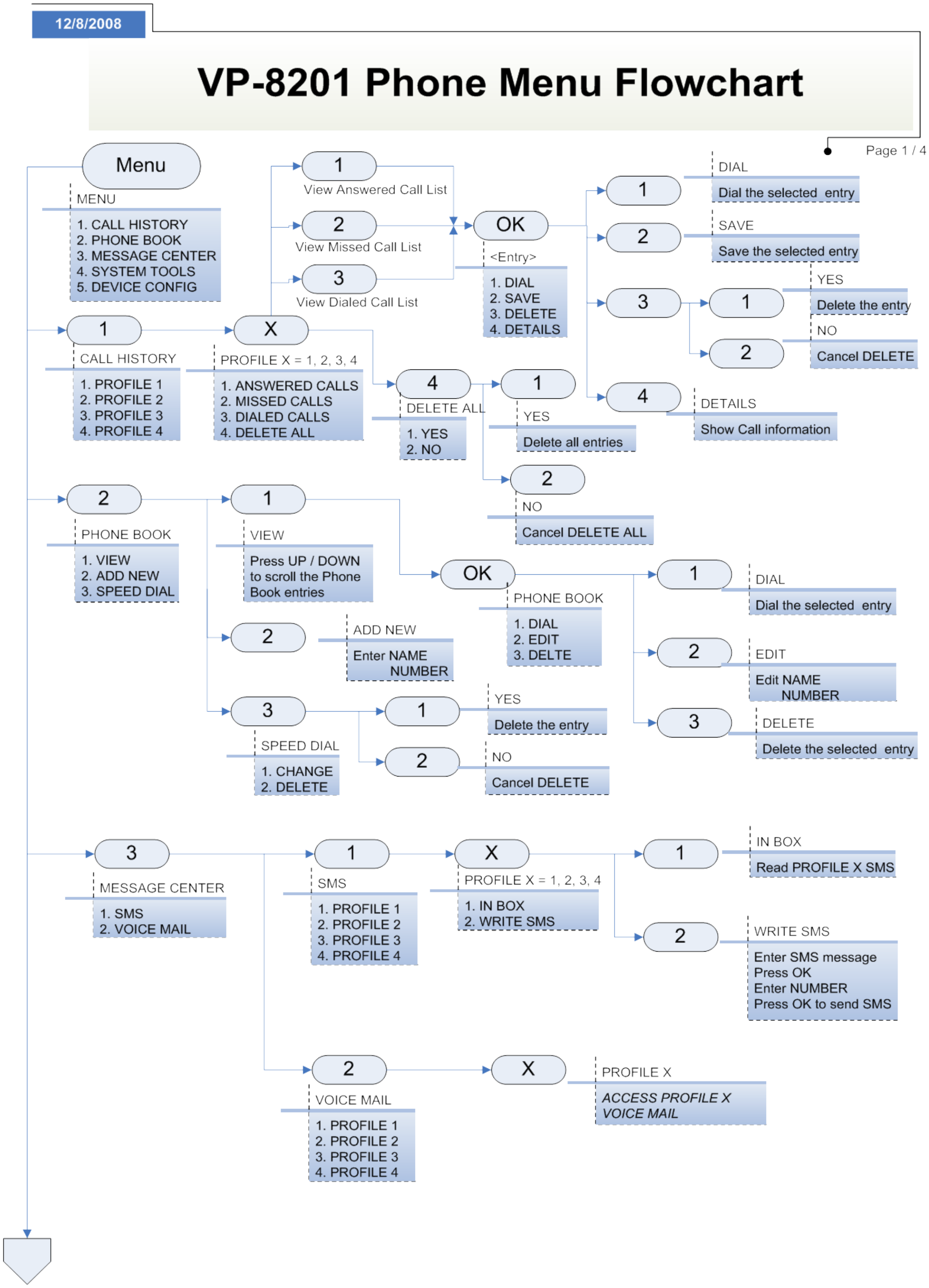
When entering alphanumeric characters and special characters, you press a key multiple times to select the desired alphanumeric characters and special characters. If the same key is not pressed after 1.5 seconds, the displayed alphanumeric character or special character will be selected as your entry.

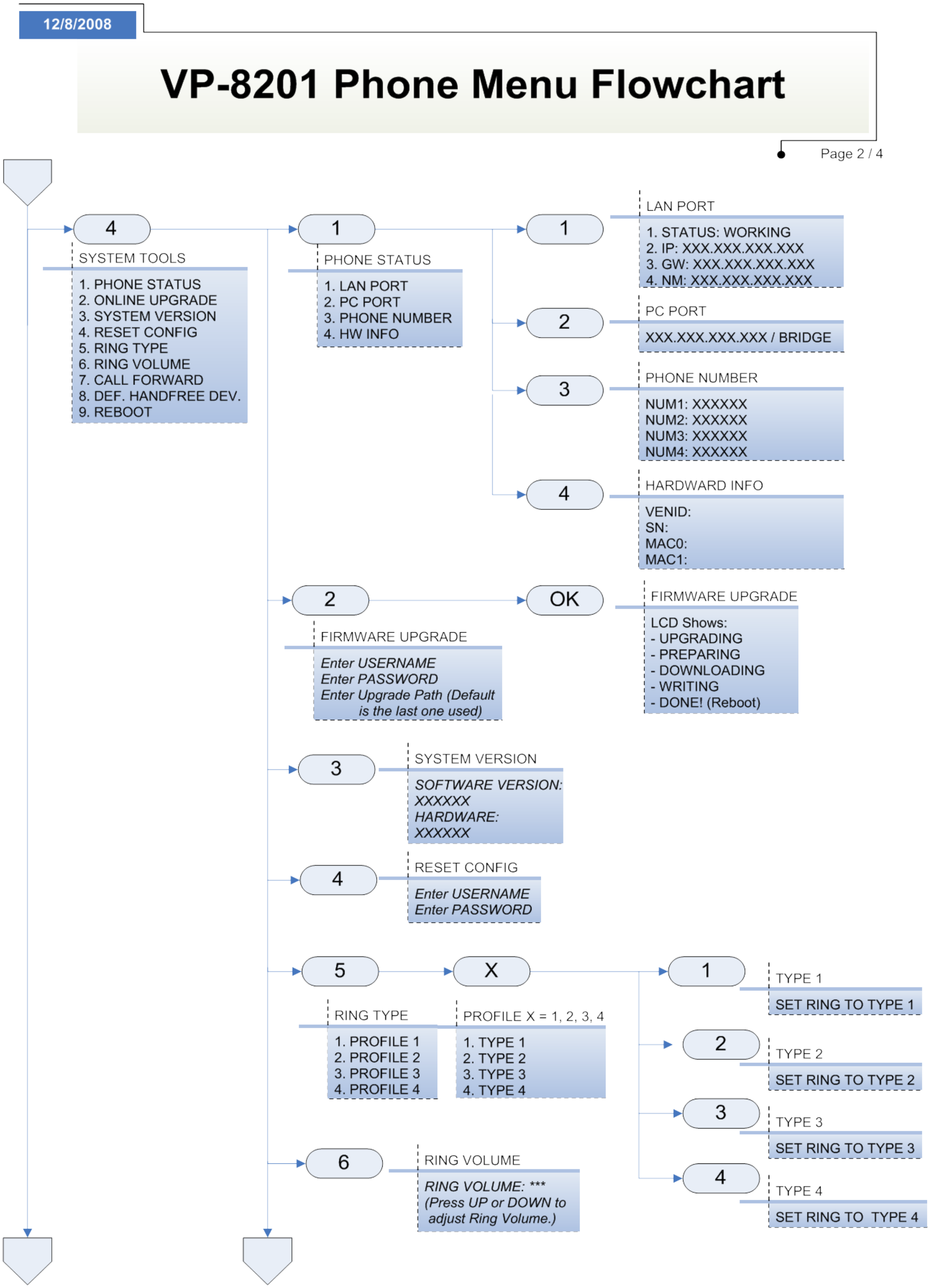
Table 2 shows the encoding scheme for the available alphabets, numbers, and symbols.

Key	Description
1	1
2	2 a b c A B C
3	3 d r f D E F
4	4 g h i G H I
5	5 Jj k l K L
6	6 m n o M N O
7	7 p q r s P Q R S
8	8 t u v T U V
9	9 w x y z W X Y Z
#	# @ % & / ~ $ [ ] { }
0	0
*****	* . , ! ? : ; ( ) blank

Table B-1 VoIP Phone Keypad Encoding Scheme

Appendix C: Phone Menu – Multiple Server Mode

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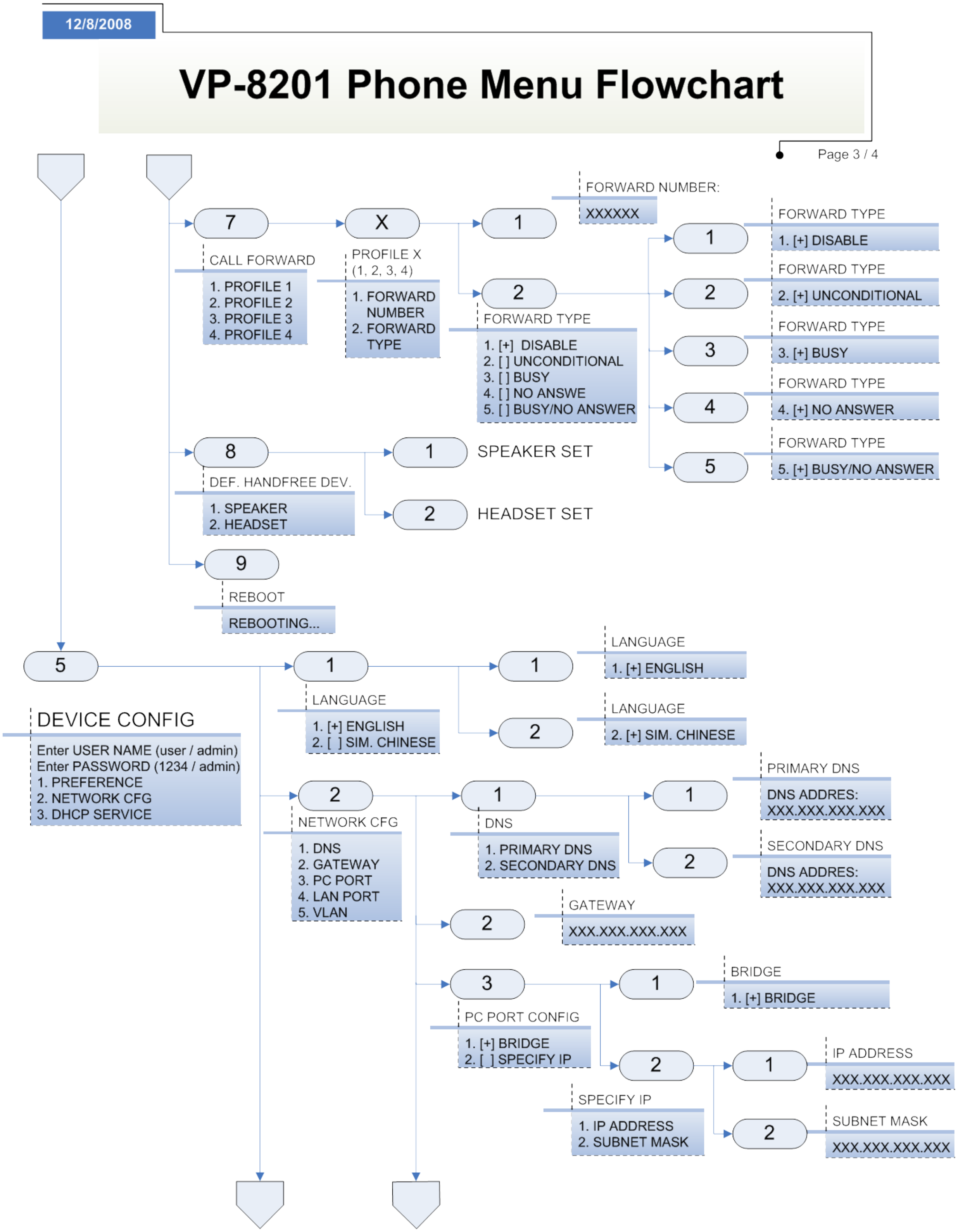
4
SYSTEM TOOLS
1. PHONE STATUS
2. ONLINE UPGRADE
3. SYSTEM VERSION
4. RESET CONFIG
5. RING TYPE
6. RING VOLUME
7. CALL FORWARD
8. DEF. HANDFREE DEV.
9. REBOOT
1
PHONE STATUS
1. LAN PORT
2. PC PORT
3. PHONE NUMBER
4. HW INFO
1
LAN PORT
1. STATUS: WORKING
2. IP: XXX.XXX.XXX.XXX
3. GW: XXX.XXX.XXX.XXX
4. NM: XXX.XXX.XXX.XXX
2
PC PORT
XXX.XXX.XXX.XXX / BRIDGE
3
PHONE NUMBER
NUM1: XXXXXX
NUM2: XXXXXX
NUM3: XXXXXX
NUM4: XXXXXX
4
HARDWARD INFO
VENID:
SN:
MAC0:
MAC1:
2
OK
FIRMWARE UPGRADE
Enter USERNAME
Enter PASSWORD
Enter Upgrade Path (Default is the last one used)
FIRMWARE UPGRADE
LCD Shows:
- UPGRADING
- PREPARING
- DOWNLOADING
- WRITING
- DONE! (Reboot)
3
SYSTEM VERSION
SOFTWARE VERSION: XXXXXX
HARDWARE: XXXXXX
4
RESET CONFIG
Enter USERNAME
Enter PASSWORD
5
X
RING TYPE
1. PROFILE 1
2. PROFILE 2
3. PROFILE 3
4. PROFILE 4
PROFILE X = 1, 2, 3, 4
1. TYPE 1
2. TYPE 2
3. TYPE 3
4. TYPE 4
1
TYPE 1
SET RING TO TYPE 1
2
TYPE 2
SET RING TO TYPE 2
3
TYPE 3
SET RING TO TYPE 3
4
TYPE 4
SET RING TO TYPE 4
6
RING VOLUME
RING VOLUME: ***
(Press UP or DOWN to adjust Ring Volume.)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7
CALL FORWARD
1. PROFILE 1
2. PROFILE 2
3. PROFILE 3
4. PROFILE 4
X
PROFILE X
(1, 2, 3, 4)
1. FORWARD NUMBER
2. FORWARD TYPE
1
FORWARD NUMBER:
XXXXXX
2
FORWARD TYPE
1. [+] DISABLE
2. [ ] UNCONDITIONAL
3. [ ] BUSY
4. [ ] NO ANSWE
5. [ ] BUSY/NO ANSWER
1
FORWARD TYPE
1. [+] DISABLE
2
FORWARD TYPE
2. [+] UNCONDITIONAL
3
FORWARD TYPE
3. [+] BUSY
4
FORWARD TYPE
4. [+] NO ANSWER
5
FORWARD TYPE
5. [+] BUSY/NO ANSWER
8
DEF. HANDFREE DEV.
1. SPEAKER
2. HEADSET
1
SPEAKER SET
2
HEADSET SET
9
REBOOT
REBOOTING...
5
DEVICE CONFIG
Enter USER NAME (user / admin)
Enter PASSWORD (1234 / admin)
1. PREFERENCE
2. NETWORK CFG
3. DHCP SERVICE
1
LANGUAGE
1. [+] ENGLISH
2. [ ] SIM. CHINESE
1
LANGUAGE
1. [+] ENGLISH
2
LANGUAGE
2. [+] SIM. CHINESE
2
NETWORK CFG
1. DNS
2. GATEWAY
3. PC PORT
4. LAN PORT
5. VLAN
1
DNS
1. PRIMARY DNS
2. SECONDARY DNS
1
PRIMARY DNS
DNS ADDRES:
XXX.XXX.XXX.XXX
2
SECONDARY DNS
DNS ADDRES:
XXX.XXX.XXX.XXX
2
GATEWAY
XXX.XXX.XXX.XXX
3
PC PORT CONFIG
1. [+] BRIDGE
2. [ ] SPECIFY IP
1
BRIDGE
1. [+] BRIDGE
2
SPECIFY IP
1. IP ADDRESS
2. SUBNET MASK
1
IP ADDRESS
XXX.XXX.XXX.XXX
2
SUBNET MASK
XXX.XXX.XXX.XXX

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